SUBCOURSE SS0333 | EDITION 8

TELEPHONE SYSTEM CHARACTERISTICS

AIPD

READINESS/ PROFESSIONALISM THRU GROWTH

THE ARMY INSTITUTE FOR PROFESSIONAL DEVELOPMENT

ARMY CORRESPONDENCE COURSE PROGRAM

UNITED STATES ARMY
SIGNAL SCHOOL
FORT GORDON, GEORGIA

SIGNAL SUBCOURSE 333

TELEPHONE SYSTEM CHARACTERISTICS

SSO 333 8

This is a Signal School publication.

*** IMPORTANT NOTICE ***

THE PASSING SCORE FOR ALL ACCP MATERIAL IS NOW 70%.

PLEASE DISREGARD ALL REFERENCES TO THE 75% REQUIREMENT.

EDITION 8
8 CREDIT HOURS

SIGNAL SUBCOURSE 333
TELEPHONE SYSTEM CHARACTERISTICS

INTRODUCTION

The quality of a communication system depends on many things.

Can we recognize the voice of the speaker?

How far apart are the telephones?

What quality of reproduction is needed to enable us to communicate by voice?

What is the frequency response of the telephone receiver?

Why do we sometimes select a four-wire circuit instead of a two-wire circuit?

How does echo interfere with our conversation?

How much noise can we tolerate and still hear the message?

These technical implications refer to telephone system characteristics. Overall system characteristics are determined by the total effect of all components that make up the system. Circuit conditioners must be aware of telephone system characteristics and their effects, because many times these technicians are called upon to modify the characteristics to accommodate other types of signals such as data.

This subcourse will provide you with an introduction to telephone systems, and will present them in terms of signal levels.

This subcourse consists of four lessons and an examination, as follows:

You are urged to finish this subcourse without delay; however, there is no specific limitation on the time you may spend on any lesson.

Materials furnished: Subcourse Booklet

LESSON 1

INTRODUCTION TO TELEPHONE SYSTEMS

SCOPE............................Survey of terms and definitions used to describe characteristics of telephone communication systems; effect of these characteristics on transmission losses.

CREDIT HOURS.....................2

TEXT ASSIGNMENT..................Attached Memorandum, para 1-1 thru 1-3; Appendix A, para 1 thru 31; Appendix B, para 1 thru 9

MATERIALS REQUIRED...............None

SUGGESTIONS......................Study Appendix A and Appendix B first, and then the Attached Memorandum.

LESSON OBJECTIVES

When you have completed this lesson, you should:

1. Know that the interfering effect of noise depends upon its amplitude and frequency.

2. Know that loading of a telephone line reduces its attenuation over the voice-frequency range.

3. Know that leakage in a telephone line reduces the amount of received signal power.

4. Know that a telephone line has characteristics resembling those of a low-pass filter.

5. Know that the passband of a telephone network need be no greater than the bandwidth of the signal.

6. Know that careless use of telephones by system users can introduce disturbing noise.

7. Be able to compare weighting characteristics.

8. Be able to determine resistance of a wire in a pair from the knowledge of its resistance per loop mile.

9. Be able to determine cutoff frequency of a line from its loss characteristic.

STUDY GUIDE

Appendix A and Appendix B carry related information. As you study them, relate the information contained in the two publications. For example, paragraphs 25 and 26 in Appendix A and paragraph 5 of Appendix B discuss the effect of capacitance in a transmission line; however, the latter publication expands on the information in the former. Follow this procedure when answering the exercises in this lesson.

ATTACHED MEMORANDUM

1-1. IMPAIRMENTS TO TELEPHONIC COMMUNICATION

Interfering electrical energy which falls within the voice-frequency band (approximately 200 to 3,000 Hz) impairs the message reception of the listener in a telephone communication system. Telephone system components, as well as the end instruments, must have sufficient quality for the voice to be recognized in normal telephone communications. There is no need to have higher quality in the system than the telephone receiver can reproduce. Higher quality requires the use of a wider passband. An increase in the passband of the system invites the entry of noise and crosstalk into the signal through the unused portion of the band. Although the increased bandwidth will not require additional power for normal voice transmission, usefulness of the received signal will not be improved because the end instruments limit the bandwidth. Furthermore, greater bandwidth will require increasingly complex circuitry and apparatus which will not markedly improve voice quality. The entire emphasis on maintaining voice quality is therefore centered on minimizing the effects of impairments in the limited bandwidth required for voice transmission.

a. Responsibilities of Installers and Users. A telephone communication system is an extensive combination of wires, apparatus, and people. Some people are responsible for installation and maintenance of the system, while the users are responsible for proper operation over the system. Correct installation will minimize the introduction of impairments. Correct operation will gain for the user the best advantages offered by quality installation.

(1) Installing telephone facilities requires careful work, and testing after work has been completed. This involves good planning, layout, and workmanship. Poor workmanship in the installation process makes the work of the technical controller and circuit conditioner much more difficult than it need be. Although they cannot correct mistakes in installation, circuit controllers must be able to recognize the effect produced by those mistakes and to report their observations to higher authorities.

(2) Correct installation requires consideration of the location of power lines because the power carried over them is so much greater than the voice signal passing through telephone lines. Induced voltages from power lines may be so much greater than the voice signal as to completely overwhelm the weak telephone currents. Telephone lines should be installed so as to pass under power lines at right angles.

The small amount of power induction resulting in short lines within buildings where telephones are installed is insignificant compared with the amount picked up by the long outdoor lines.

(3) Improper selection of cable pairs can cause crosstalk between circuits. The effect of crosstalk is the same as noise, in that the disturbance impairs understandability of the received message. The immediately noticeable effect of crosstalk in a telephone system is the unintelligible conglomeration of speech sounds called "babble" coming from a large number of sources. Moreover, poor electrical connections, together with inferior quality of equipment maintenance along the cable, contributes to the amount of electrical interfering noise other than that induced by natural causes.

(4) Users of telephone systems can also impair reception by carelessness. Background noise entering the telephone microphone has the same effect on distant reception as an equivalent amount of electrical noise introduced by extraneous sources. Moreover, the sidetone path through the telephone can introduce some of the noise picked up by the microphone into the local earpiece. A firm voice and clear pronunciation also help improve understanding of the received voice. The ability to understand the message is, in the final analysis, the determinant of successful communication.

b. Disturbing Effect of Noise. The disturbing effect of noise to a listener depends on its volume (amplitude) and pitch (frequency). A series of tests were made with many people to find the frequencies that interfered most with their listening. The results are shown in figure 1-1. Note that the disturbing effect peaks up rather sharply in the neighborhood of 1,100 Hz; that is, interfering frequencies near 1,100 Hz are the most important so far as telephone communication is concerned.

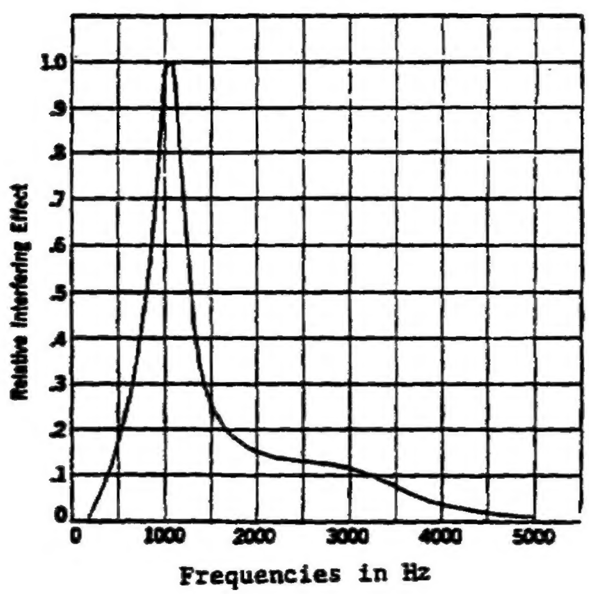


Figure 1-1. Relative interfering effect of noise at different frequencies.

1-2. LINE OR CIRCUIT WEIGHTING

a. Weighting Networks. The interfering effect of noise on a listener varies with both the relative loudness and the pitch (power and frequency). Therefore, the importance of the components of noise at the different frequencies within the voice band must be taken into consideration in determining the total amount of interference. Moreover, the interfering effect varies according to the sensitivity of the receiving device that converts the noise currents into audible sound. For these reasons, in measuring noise on telephone networks, it is desirable to employ weighting networks. These networks integrate the noise power over the voice-frequency range by giving each small band of frequencies a weighting proportional to its contribution to the total interfering effect. Different weighting networks may be used to simulate the effect of different telephone networks.

b. Line Weighting Curves. A line weighting curve (sometimes called circuit weighting curve) includes the attenuating effect of a typical telephone network, including the exchange, subscriber loop, and telephone set. The weighting curve of a telephone receiver alone is somewhat different from the line weighting curve. The line weighting curve is the more important one to the circuit conditioner because it more closely approximates the average operating characteristic of a complete telephone network.

(1) Typical line weighting curves are shown in figure 1-2. Designations F1A, 144, and C-message weighting are terms applied to the characteristics of telephone circuits using selected telephone handset receiving units. The C-message weighting curve is the most important one to the circuit conditioner because it is the most widely used.

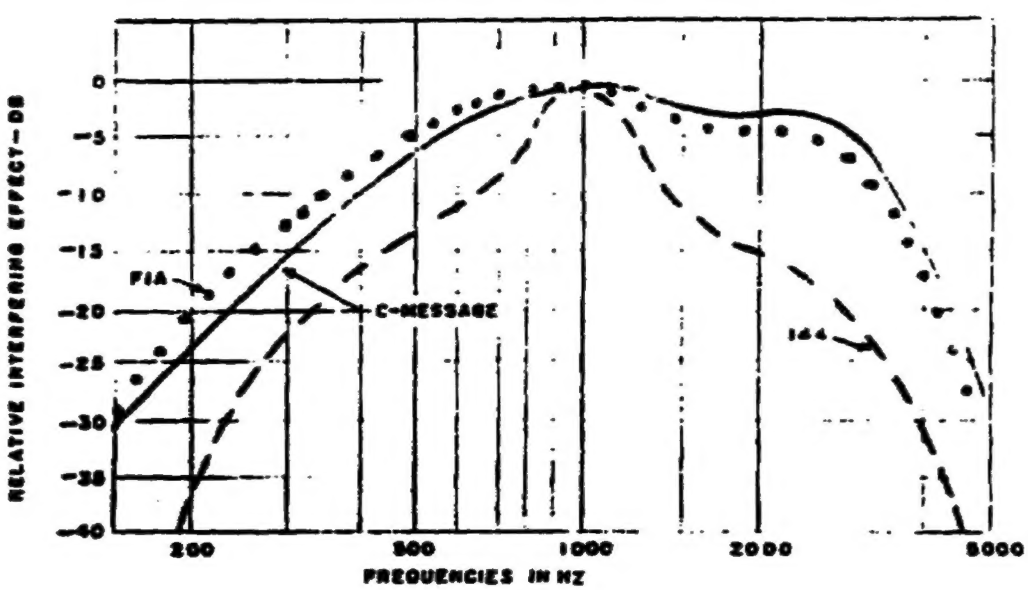


Figure 1-2. Line weighting curves.

(2) All three curves are plotted so that 0 db at 1,000 Hz is the minimum attenuation each of the three telephone receivers introduces. A set of sensitivity curves would indicate C-message having the highest sensitivity, F1A the next, and 144 the least. C-message weighting showing the widest bandwidth and highest sensitivity will pass both signal and noise with less attenuation than the other two weightings. The improvement in the telephone receiver voice quality requires a similar reduction in circuit noise. Noise measurements are usually taken with respect to 1,000 Hz because all three weighting curves show minimum loss (maximum output) at near that frequency. It is apparent from these curves that a listener will hear maximum sound from the handset receiver in the neighborhood of 1,000 Hz. It is also apparent that C-message weighting delivers a range of frequencies with more uniform response than either F1A or 144 weightings.

1-3. AMPLITUDE-FREQUENCY DISTORTION

Unequal attenuation is called amplitude-frequency distortion because the amplitude of the received signal changes as the frequency of the transmitted signal is changed. The measurement technique requires a technician to measure and record received signal levels at specified frequencies. The transmitting station sends the specified frequencies at constant level. The difference between the received signal and the constant level indicates the amount of amplitude-frequency distortion present in the line or circuit. Amplitude-frequency distortion is caused by the phase shift between current and voltage through a reactive circuit. Values of L and C in the line, together with the frequency of the signal, determine the effective value of reactance that exists at any particular time.

a. True and Apparent Power.

(1) True power. True power is the power in a circuit when the load is purely resistive. This condition is obtained when the phase angle between current and voltage is zero. At this time, power is the product of voltage and current.

(2) Apparent power. Apparent power is the power in a circuit when the load is reactive. Reactance due to inductance and capacitance in the load causes a shift in the phase angle between current and voltage. A change in reactance occurs with each change of frequency, causing the power in the load to vary with frequency. Thus, when we measure the output level from a telephone line as the transmitted signal frequency is varied, we will notice a change in measured power output.

b. Reactance Variation. Inductive and capacitive reactances in a telephone line do not change in the same proportion to a given change in frequency. When loading coils are inserted in the line, this effect is even more pronounced. When a loaded telephone line transmits frequencies higher than cutoff, distortion will greatly increase beyond that point. If frequencies higher than the cutoff must be transmitted over this line, better results will be achieved by removing the loading coils and making correction in a suitable terminating device such as an equalizer.

c. Loading and Data Transmission. Loading consists of inserting values of inductance in the line to reduce attenuation. However, you know from studies of electrical fundamentals that inductance tends to oppose changes in current. Although adding L to reduce amplitude-frequency distortion is a practical solution to one problem, the L in the loading coils causes another problem if high-speed data are to be sent over the line. The higher frequencies which are part of the data signal will be greatly affected, causing so much envelope-delay distortion as to possibly make the received data signal nearly useless. As a general rule, telephone circuits will perform better in data communication if they are not loaded.

STUDY EXERCISES

In each of the following exercises, select the ONE answer that BEST completes the statement or answers the question. Indicate your solution by circling the letter opposite the correct answer in the subcourse booklet.

1. A transmitter circuit in a telephone has two power sources: voice power and battery power. In the circuit of figure 3, Appendix A, the transmitter controls the output power by

a. applying the ac voice signal directly to the line.

b. varying the resistance in series with the battery.

c. varying the direct current in the secondary circuit.

d. rectifying the voice current to provide dc in the primary circuit.

2. When a standard level of voice signal power is dissipated in a 600-ohm precision resistor, the standard voltage of 0.775 is developed. These conditions produce the standard voice power, which is

a. 1 milliwatt.

b. 0.1 picowatt.

c. 0.01 microwatt.

d. 0.001 milliwatt.

3. Circuit conditioners must know the effects of attenuation on a telephone line. Attenuation on a telephone line causes

a. noise level to rise.

b. voice power to be reduced.

c. induced voltages to appear.

d. crosstalk to interfere with communication.

4. The amplitude-frequency response of a loaded line is said to be "flat" when attenuation is constant over a given frequency range. The attenuation curve in figure 6 of Appendix B approaches a flat response over the voice-frequency range extending between

a. 300 Hz and 3,500 Hz.
c. 3,500 Hz and 3,800 Hz.

b. 300 Hz and 4,000 Hz.
d. 3,800 Hz and 4,500 Hz.

5. A term often used by designers of long-distance trunk circuits is <u>attenuation constant</u>. This is a mathematical expression which combines the effect of the four properties of a telephone line, namely,

a. resistance, leakage, capacitance, and inductance.

b. leakage, capacitance, inductance, and conductance.

c. inductance, conductance, resistance, and leakage.

d. capacitance, inductance, conductance, and resistance.

6. Assume that two telephones 20 miles apart are connected by a wire pair that has 74 ohms resistance per loop mile (37 ohms per wire). If you measure the total resistance in the wire with the far end shorted (looped back), the ohmmeter should read approximately

a. 370 ohms.
c. 1,480 ohms.

b. 740 ohms.
d. 2,960 ohms.

7. The amount of leakage in telephone cables is usually small because the

a. metal sheath is ungrounded.

b. cable pairs are protected from moisture.

c. insulation resistance between wire pairs is infinite.

d. telephone cable is supported by messenger cable rather than by crossarm insulators.

8. Unequal attenuation of a telephone line is caused by its electrical properties. As the frequency rises, the electrical properties that cause less signal current to arrive at the receiver are

a. inductance and capacitance.
c. resistance and leakage.

b. capacitance and resistance.
d. leakage and inductance.

9. Assume that a telephone line circuit has the equivalent inductance of 6 millihenries and capacitance of 1 microfarad. If the resistance of the line is 157 ohms, the approximate impedance of the circuit at 2,000 Hz is

a. 37.6 ohms.

b. 79.0 ohms.

Hint: Refer to tables I and II in Appendix A.

c. 113 ohms.

d. 157 ohms.

10. Complete cancellation of capacitive reactance by inductive reactance in a telephone line is not desirable because

a. howling or singing may result from resonance.

b. line current drops suddenly at the frequency of resonance.

c. line impedance rises to a high value, attenuating the signal.

d. the sharply increased loss at resonance causes distortion in the voice signal.

11. When we load a telephone line to reduce attenuation, we add

a. inductance in series with the line.

b. capacitance in series with the line.

c. inductance in parallel with the line.

d. capacitance in parallel with the line.

12. The loading curve in figure 6 of Appendix B shows variation in attenuation of a transmission line at different frequencies. What effect does this loading have on voice power as the signal passes through the line?

a. The attenuation increases sharply above cutoff, reducing voice power above this frequency.

b. The attenuation increases sharply below cutoff, reducing voice power below this frequency.

c. The attenuation decreases sharply above cutoff, allowing more voice power to pass.

d. The attenuation decreases sharply below cutoff, allowing more voice power to pass.

13. The function of a repeater amplifier in a telephone system is to

a. introduce loss.

b. reduce distortion.

c. overcome attenuation.

d. eliminate the need for loading.

14. Telephone repeaters include facilities to change two-wire circuits to four-wire inside the repeaters. This is made necessary by the fact that repeaters normally have

a. bidirectional amplifiers that raise signal level the same amount in each direction of transmission.

b. series-connected amplifiers so as to reinforce feedback in the loop from one telephone transmitter to its receiver.

c. two amplifiers in cascade because one amplifier seldom has sufficient gain to overcome line loss.

d. two amplifiers, one for each direction of transmission.

15. Using a 4-kHz circuit to pass frequencies of 0.3 to 3 kHz may be detrimental to voice communication because

a. more power will be needed for the same distance.

b. the added complexity of equipment increases cost.

c. all end instruments must be changed to accommodate the increased bandwidth.

d. the unused portion of the baud allows passage of additional noise and crosstalk.

16. Careful installation and maintenance of a telephone system can be negated by carelessness of the user. How can the user improve the quality of the received voice?

a. Shield the microphone and the earpiece against background noise.

b. Speak with a firm voice and clear pronunciation, and shield the microphone against background noise.

c. Shield the earpiece against background noise, and keep the telephone away from power lines.

d. Keep the telephone away from power lines, and speak with a firm voice and clear pronunciation.

17. The relative interfering effect of noise depends on its frequency, as shown in figure 1-1. If the interference is produced by four tones of a telegraph carrier terminal the tone that will have the greatest interfering effect is

a. 425 Hz.

b. 765 Hz.

c. 1,155 Hz.

d. 2,795 Hz.

18. Assume that a transmission test set uses the three weighting networks shown in figure 1-2. If the received level at 1,000 Hz is adjusted to 0 dbm, the weighting levels at 1,500 Hz will be approximately

	F1A	144	C-message
a.	-3 dbm	-12 dbm	-2 dbm
b.	-8 dbm	-2 dbm	-12 dbm
c.	-2 dbm	-8 dbm	-12 dbm
d.	-12 dbm	-2 dbm	-8 dbm

19. The most commonly used weighting characteristic in modern telephone equipment is C-message. Compared with F1A and 144 weightings, C-message has a

a. narrower bandpass.

b. greater loss at 1,000 Hz.

c. more uniform frequency response.

d. sharper peak response ac 1,000 Hz.

20. When a telephone line is to carry high-speed data signals, the line circuit may have to be modified by removing

a. repeaters.

b. amplifiers.

c. attenuators.

d. loading coils.

CHECK YOUR ANSWERS WITH LESSON SOLUTION SHEET (PGS 59 and 60) AND MAKE NECESSARY CORRECTIONS.

LESSON 2

NETWORK LOSSES AND GAINS

SCOPE............................Determination of transmission losses and gains; calculation of signal-to-noise ratios; stating circuit impairment levels.

CREDIT HOURS......................2

TEXT ASSIGNMENT...................Attached Memorandum, para 2-1 thru 2-6

MATERIALS REQUIRED................None

SUGGESTIONS.......................None

LESSON OBJECTIVES

When you have completed this lesson, you should:

1. Be able to convert dbm to milliwatts, and vice versa.

2. Be able to determine transmission losses and gains from a transmission level diagram.

3. Be able to specify noise level in relative terms.

4. Be able to calculate signal-to-noise ratio.

5. Know that the maximum loss permitted by the Military Transmission Plan is 36 db.

ATTACHED MEMORANDUM

2-1. DECIBELS AND THEIR USE

Decibels and their derivatives are the basic tools of the circuit conditioner. Host of the parameters specified by the Defense Communications Agency (DCA) are related to the decibel. The primary reason for using decibels is to make calculations of quantities simple and quick. Instead of multiplying or dividing large numbers, it is only necessary to add or subtract decibels to arrive at useful answers. This simplicity results from the logarithmic nature of the decibel formula:

$$db = 10 \log_{10} P_1/P_2 = 20 \log_{10} E_1/E_2 = 20 \log_{10} I_1/I_2.$$

<u>a</u>. <u>Ratios</u>. A simplified table showing power, voltage, and current ratios is shown below. Each ratio assumes equal values of impedance. This is one of the reasons that most test sets and circuits use standardized impedances

(mostly 600 ohms, with some at 135 ohms). It is important to note that nowhere in table I is an actual level of power indicated. This emphasizes the fact that a decibel represents a ratio, not a power level. Although not shown in the table, a ratio of 1/1 is 0 db, since there is neither gain nor loss.

b. Reference Levels. Both the telephone industry and military communicators have standardized all sound level measurements on 1 milliwatt (1×10^{-3} watt), assigning the symbol: 0 db - 1 mw in 600 ohms. Other industries connected with sound transmission, such as broadcast, noise abatement, stereo components, etc., assign other values or standards. A few are illustrated below.

0 db6m = 6 milliwatts (6×10^{-3} watt)

0 db12m = 12 milliwatts (12×10^{-3} watt)

0 dbw = 1 watt

0 dbrn = 1 picowatt (1×10^{-6} watt)

c. Instrumentation. All test instruments must carry some indication of the reference used. Otherwise level readings have little useful significance. Unfortunately, most level-measuring instruments show DB on the meter faces and are therefore called decibel meters. However, if you look closely you may see in fine print on the meter face an expression similar to "0 db = 1 mw in 600 ohms." Such a meter actually reads levels in dbm referred to its calibration at 1 mw. If a measured level is given to a circuit conditioner simply as so many db's, as is so often the case, he should immediately inquire as to the reference level used by the instrument.

d. Rules for Use of Decibels.

(1) A decibel value may be either added to or subtracted from another decibel value to indicate either gain or loss respectively.

(2) A decibel value may be either added to or subtracted from a level value in a circuit to indicate the signal level resulting from the gain or loss.

(3) Decibel figures must never be multiplied or divided.

(4) A level value must NOT be added to another level value. To do so is the equivalent of multiplying power instead of adding it.

(5) Examples of correct and incorrect usage are given below.

Example 1: Assume that a circuit contains three amplifiers and an attenuator. If the gain of the first amplifier is 5 db, the second is 12 db, and the third 17 db, and the attenuator loss is 20 db, what is the overall gain of the combination circuit?

Add 5, 12, and 17, and subtract 20.

5 + 12 + 17 - 20 = +14 db, a gain of 14 db.

TABLE I

POWER, VOLTAGE, AND CURRENT RATIOS

NUMBER OF DECIBELS	POWER RATIO	VOLTAGE (CURRENT) RATIO	NUMBER OF DECIBELS	POWER RATIO	VOLTAGE (CURRENT) RATIO
0.5	1.122	1.059	12.5	17.78	4.217
1.0	1.259	1.122	13.0	19.95	4.467
1.5	1.413	1.189	13.5	22.39	4.732
2.0	1.585	1.259	14.0	25.12	5.012
2.5	1.778	1.334	14.5	28.18	5.309
3.0	1.995	1.413	15.0	31.62	5.623
3.5	2.239	1.496	15.5	35.48	5.957
4.0	2.512	1.585	16.0	39.81	6.310
4.5	2.818	1.679	16.5	44.67	6.683
5.0	3.162	1.778	17.0	50.12	7.079
5.5	3.548	1.884	17.5	56.23	7.499
6.0	3.981	1.995	18.0	63.10	7.943
6.5	4.467	2.113	18.5	70.79	8.414
7.0	5.012	2.239	19.0	79.43	8.913
7.5	5.623	2.371	19.5	89.13	9.441
8.0	6.310	2.512	20	10^2	10.00
8.5	7.079	2.661	30	10^3	3.162×10
9.0	7.943	2.818	40	10^4	10^2
9.5	8.913	2.985	50	10^5	3.162×10^2
10.0	10.00	3.162	60	10^6	10^3
10.5	11.22	3.350	70	10^7	3.162×10^3
11.0	12.59	3.548	80	10^8	10^4
11.5	14.13	3.758	90	10^9	3.162×10^4
12.0	15.85	3.981	100	10^{10}	10^5

Example 2: Assume that the input level to the circuit in Example 1 is -3 dbm. What is the output level from the circuit?

Add +14-db gain to -3-dbm level to find the output level.

+14 + (-3) = +11 dbm.

e. Uniform Channel Levels. Equal levels are always desirable from each of the several channels in a frequency-division-multiplexed (FDM) line signal. In most cases, this line signal passes through modulators or amplifiers. The channel having the strongest of several signals will predominate at the expense of the other channels. The result is an accentuated difference in level of the various frequencies making up the signal, together with possible amplifier overloading.

f. Maximum Undistorted Output Levels. The maximum undistorted output level of an amplifier must be observed to prevent distortion. In other words, the amplifier must not be overloaded. For example, if the maximum undistorted power output of an amplifier is +10 dbm, a technician should not decrease the attenuator loss in an attempt to raise the output level above the maximum permissible amount. He must add an additional amplifier. Failure to follow this precaution results in amplitude distortion in case of a single-tone or a voice signal, and intermodulation distortion in case of an FDM signal.

2-2. ABSOLUTE VS RELATIVE LEVELS

The Defense Communications Agency specifies level parameters for the Defense Communications System based on absolute and relative quantities.

a. Absolute Levels. Absolute levels are the actual measured levels and represent existing power levels. Power levels are indicated as a number of decibels above or below a given power level. Thus, dbm has a reference value of 1 milliwatt (0 dbm). Likewise, dbw has a reference value of 1 watt (0 dbw). The expression dbw is seldom used in communication because of the large power involved, but it is widely used in connection with output power of radio transmitters. Power levels can become very low in communication. For example, the average human ear can hear noises as low as 1×10^{-6} of 1 milliwatt. Test sets designed to measure noise are calibrated with respect to -90 dbm, which represents an absolute level of 1×10^{-9} milliwatt. A graph relating power in milliwatts to dbm is shown in figure 2-1. Note that every change of power in the ratio of 1/2 or 2/1 results in a 3-db change of level in dbm. Likewise, a change of 1/4 or 4/1 gives a 6 db change of level in dbm.

b. Voltage and Power. Power can be measured by test sets either in terms of actual power or in terms of voltage. The relationship between voltage and power is indicated in table II. Note that power development does not depend on voltage alone. The most important consideration is the combination of measured voltage in a specified value of impedance. The standard when using a voltmeter for determining power level is: 0 dbm = 0.775 volts in an impedance of 600 ohms. In 150 ohms impedance, approximately 0.39 volt produces 0 dbm (1 mw). In 75 ohms, 0.274 provides 0 dbm. Working the other way in table II, a level of -3 dbm produces a voltmeter reading of 0.548 in a 600-ohm circuit.

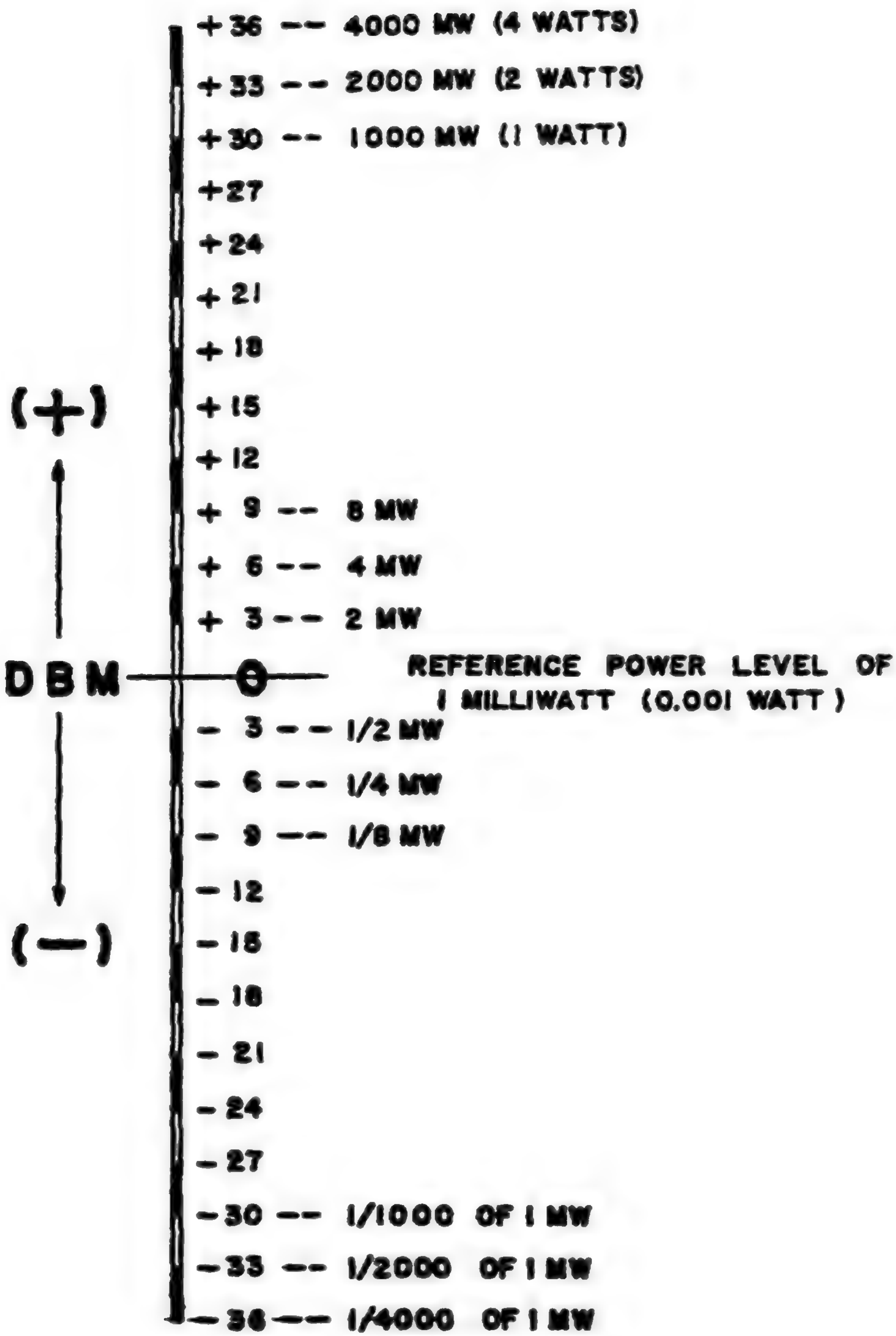


Figure 2-1. Approximate power levels.

TABLE II

VOLTAGE VARIATION

1,000 Hz Sine Wave Test Tone (dbm)	RMS Volts 600-Ohm Load	RMS Volts 150-Ohm Load	RMS Volts 75-Ohm Load
10	2.45	1.225	0.8660
9	2.18	1.090	0.7719
8	1.95	0.937	0.6879
7	1.74	0.866	0.6132
6	1.55	0.772	0.5465
5	1.40	0.688	0.4870
4	1.23	0.614	0.4340
3	1.10	0.547	0.3869
2	0.977	0.487	0.3448
1	0.870	0.434	0.3073
$\boxed{0}$ STANDARD	$\boxed{0.775}$	0.3873	0.2739
-1	0.697	0.348	0.2441
-2	0.616	0.308	0.2175
-3	0.548	0.274	0.1939
-4	0.489	0.244	0.1728
-5	0.436	0.218	0.1540
-6	0.388	0.194	0.1373
-7	0.347	0.173	0.1223
-8	0.309	0.154	0.1090
-9	0.276	0.138	0.09718
-10	0.245	0.1225	0.08660

c. Relative Power Levels. Relative power levels are not measured, but they are stated in terms of the power that exists, not a standard reference power level. However, since the power that exists is referred back to 1 mw, relative power levels must indicate that they refer back to some other selected power level. The expression 0 dbm0 indicates that this is a level to which all others will be referred in a given circuit. The expression does not indicate the actual power, but that the power selected for reference does eventually refer back to 1 milliwatt. Remember--relative power levels are calculated, not measured; they are calculated with respect to a selected level which is declared the reference. The expression 0 dbm0 has value only when the existing power level selected for reference is stated. That point in the circuit where the selected reference power level appears is thereafter called the transmission level point (TLP).

d. Channel Signal Levels. In some situations the circuit conditioner must determine the level of each channel signal within a composite signal from an FDM terminal. He can easily arrive at this determination by a simple calculation. Normally, he cannot measure the level of each channel signal because it is enclosed in the "bundle" of signals that travel the line together. However, before he makes the calculation he should know the measured level of the composite signal, and he should know the number of channels in use. Further, he has to assume that each of the various channel signals has approximately the same level.

Example 1: Assume that the measured level of an eight-channel FDM composite signal is +6 dbm. What is the level of each channel signal? The scale in figure 2-1 shows that +6 dbm represents a power level of 4 mw. Dividing 4 mw by 8 channels we find the power in each channel to be approximately 1/2 mw. Referring once again to the scale in figure 2-1, it is evident that the level in dbm for 1/2 mw of power is -3 dbm. To state this another way, the composite level of eight -3-dbm channel signals is +6 dbm.

Example 2: If the level produced by each channel of an FDM terminal is -6 dbm, what composite level can be expected for 16 channels? The scale in figure 2-1 shows that the power equivalent for -6 dbm is 1/4 mw. Multiply this by 16 to obtain 4 mw. The scale shows 4 mw is equivalent to +6 dbm.

2-3. CIRCUIT DESCRIPTION

Assume that a technical control center contains three bays through which passes a line circuit carrying a signal to equipment, as shown in figure 2-2.

a. Signal Path. The input signal arrives at bay 1 and passes through a variable-loss attenuator and a 35-db fixed-gain amplifier. The overall gain of bay 1 depends only upon the attenuator setting. The signal then passes through bay 2, which has jacks for equal-level patching. The signal then passes through bay 3, which contains pads to establish the output level needed by station equipment.

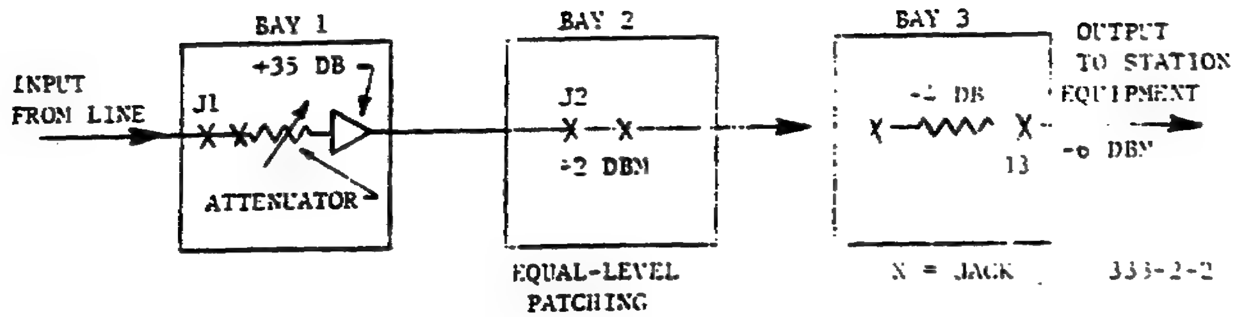


Figure 2-2. Three-bay lineup in a technical control center.

b. Circuit Specifications. The planned levels in the circuit are as follows:

(1) Expected (normal) level of input signal at bay 1 measures -12 dbm.

(2) The level at bay 2 is adjusted to -2 dbm. The overall gain of the attenuator-amplifier combination is therefore 10 db (difference between -12 dbm and -2 dbm). The level of every circuit passing through bay 2 is adjusted to the same level (-2 dbm) to make possible equal-level patching.

(3) A 4-db pad in bay 3 drops the level to -6 dbm for the equipment which the line circuit serves. This output level will always be maintained so long as the level is correctly adjusted by the attenuator in bay 1. The attenuator is therefore the only variable that need be adjusted in this line circuit.

c. Transmission Level Diagram. A transmission level diagram shows the changes in level which the signal undergoes as it passes through a circuit. Horizontal lines show levels, while vertical lines indicate amplifier gains. Slanting lines indicate losses. The transmission level diagram of the three-bay lineup in figure 2-2 is shown in figure 2-3. A horizontal line representing 0 dbm (1 mw) is always used as reference.

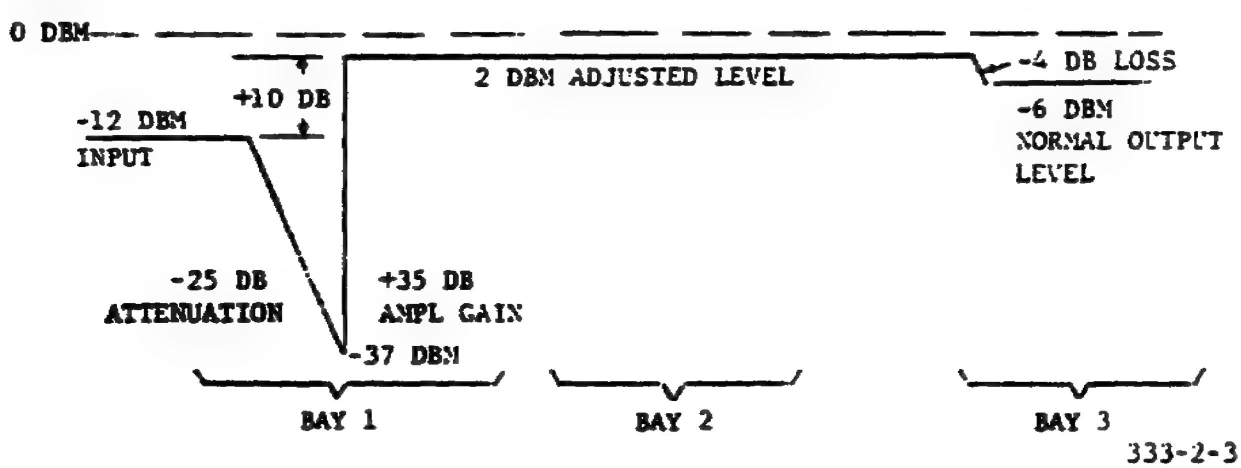


Figure 2-3. Transmission level diagram.

(1) Normal input level is -12 dbm which the attenuator drops to -37 dbm, a loss of 25 db. The 35-db fixed-gain amplifier raises the level of -37 dbm to the required -2 dbm adjusted level. The overall gain of the attenuator-amplifier combination is therefore the difference between -12 dbm and -2 dbm, an overall gain of 10 db.

(2) The signal level of -2 dbm remains without change through the jack circuits of bay 2. In other words, there is neither gain nor loss in bay 2. Another way to say this is that the gain is 0 db and the loss is likewise 0 db.

(3) A 4-db pad in bay 3 drops the signal level to -6 dbm, the level required by the equipment.

d. Loss-Gain Characteristic. A loss-gain characteristic is a graph that depicts only the changes in level without reference to levels. Figure 2-4 shows the characteristic of the three-bay layout in figure 2-2.

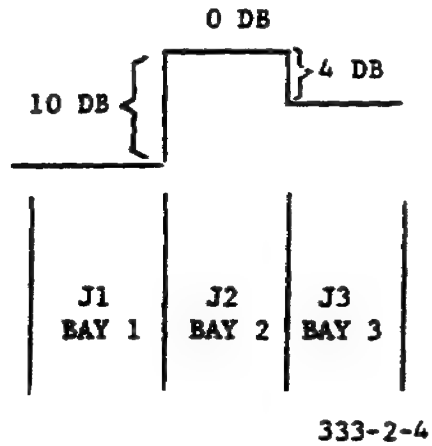


Figure 2-4. Loss-gain characteristic.

(1) The overall gain of the attenuator-amplifier combination in bay 1 is 10 db.

(2) There is neither gain nor loss in bay 2.

(3) The loss in bay 3 is 4 db.

e. Transmission Level Point. The advantage of the loss-gain characteristic is that it permits a prediction of circuit performance without resort to levels. Thus if one level is given at any point in the system, all other levels can be determined from it. The point selected becomes the TLP, to which all losses and gains in the system are thereafter referred. The TLP is assigned the reference level of 0 dbm0, and all other points in the system are identified as being either positive (gain) or negative (loss) with respect to the level at the TLP. Again, referring to the three-bay lineup shown in figure 2-2, we will derive the loss-gain characteristics of three different TLP designations. Assume that the actual (absolute) input level at jack J1 in bay 1 is -12 dbm.

f. Using the TLP.

TLP at / Levels at	J1	J2	J3
J1 Absolute Relative	-12 dbm 0 dbm0	-12 dbm -10 dbm0	-12 dbm -6 dbm0
J2 Absolute Relative	-2 dbm +10 dbm0	-2 dbm 0 dbm0	-2 dbm +4 dbm0
J3 Absolute Relative	-6 dbm +6 dbm0	-6 dbm -4 dbm0	-6 dbm 0 dbm0

Example 1: The TLP at jack J1 is taken as zero (0 dbm0). Reading downward under J1 it is evident that the relative level at J2 is +10 dbm0, and at J3 it is +6 dbm0. The actual levels at jacks J2 and J3 will therefore be -2 dbm and -6 dbm, respectively.

Example 2: Assume that the level is -2 dbm at TLP (J2) in bay 2.
What are the actual levels at jacks J1 and J3?
The relative levels at J1 and J3 will be -10 dbm0 and -4 dbm0, respectively. The actual levels at J1 and J3 will be -12 dbm and -6 dbm, respectively.

Example 3: Assume that the level is -6 dbm at TLP (J3) in bay 3.
What are the actual levels at jacks J1 and J2?
The relative levels at J1 and J2 will be -6 dbm0 and 44 dbm0, respectively. The actual levels at J1 and J2 will be -12 dbm and -2 dbm.

Example 4: Assume that the TLP is at jack J1 with normal level of -12 dbm. Since this is the TLP, the two values are related: -12 dbm = 0 dbm0. Now suppose the input signal drops to a level of -18 dbm. What is the relative level?
The level of -18 dbm is 6 db below the reference level at TLP, and is therefore quoted as -6 dbm0.

Example 5: Assume that the TLP is at J2 with normal signal level of -2 dbm. Assume further, that the maximum relative noise at this point in the circuit is established at -40 dbm0. What is the actual noise level?
40 db below -2 dbm is -42 dbm. That is the actual noise level.

2-4. CIRCUIT ADJUSTMENTS

The only adjustment in the circuit illustrated in figure 2-1 is the attenuator, which controls the overall gain of the amplifier arrangement in bay 1. The gain of the amplifier remains fixed at 35 db, and the input signal level is adjusted by the attenuator.

a. Equal-Level Patching. The equal-level patching panel in bay 2 permits circuits to be patched easily and quickly. As long as equal levels are maintained, the technician need not make any level measurements before he makes the patches. Whether the levels remain equal depends on three factors: level of the arriving input signal, gain of the amplifier, and adjustment of the attenuator. One of the duties of the technical controller is to make frequent checks of levels of incoming signals at J1. On the other hand, J3 can be used for the same purpose, because any variation of input signal level at bay 1 reflects in a similar change of level in J3.

b. Attenuator Adjustment.

Example 1: Assume that the input signal level drops from -12 dbm to -20 dbm. How must the attenuator be adjusted?
The level in bay 2 drops 8 db from -2 to -10 dbm, because the input signal level dropped the same amount. The attenuator loss must therefore be reduced 8 db so the amplifier can raise the signal level back up to the required -2 dbm in bay 2.

Example 2: Assume that the input signal level rises from -12 dbm to -6 dbm. Row must the attenuator be adjusted?
The level in bay 2 rises 6 db from -2 to +4 dbm. The attenuator loss must therefore be increased 6 db so the previous level of -2 dbm will be produced by the amplifier.

2-5. SIGNAL-TO-NOISE RATIO

The signal-to-noise ratio (S/N) in a telephone communication circuit must always be maintained at a high value. This is especially important where data communication is intended. The higher the S/N, the more satisfactory the circuit will be for communication. When S/N approaches 1/1 (0 db), noise level is as high as the signal level, making the circuit worthless for communication. An absolute minimum S/N that can be tolerated is 4/1 (6 db).

a. Noise Level Statement. Noise level can be stated in either absolute or relative terms. Each method has its advantages under different test conditions. Noise can be stated with respect to the expected level at the TLP, or it can be seated with respect to an established noise level reference.

b. Absolute Level. The absolute, or actual, level of noise existing in a circuit is measured by a test set that is calibrated in terms of a power reference. Most test equipment of this type uses -90 dbm as a zero reference noise (rn), making all quoted figures a positive value (+). Further, each test set must state the type of weighting, which affects the relative value of interfering noise. A comparison of noise reference levels used for some of the more widely used test sets is shown in the chart below.

DBM \ WTG	C-Message 1-kHz dbrn	C-message 0-3-kHz band dbrnc	F1A 1-kHz dba	F1A 0-3-kHz band dba
-80	10	8	5	2
-81	9	7	4	1
-82	8	6	3	0
-83	7	5	2	
-84	6	4	1	
-85	5	3	0	
-86	4	2		
-87	3	1		
-88	2	0		
-89	1			
-90	0			

(1) A test set reading noise in dbrn uses C-message weighting while receiving noise through a narrowband filter designed to pass 1-kHz tone.

(2) When controls are adjusted to measure dbrnc, C-message weighting is used along with a filter designed to pass noise across a band from 0-3 kHz.

(3) A test set reading noise in dba uses F1A weighting while receiving noise through the narrowband 1-kHz filter.

(4) When controls are adjusted to measure dba with F1A band weighting, the 0-3-kHz filter passes noise within its response.

(5) For example, -82 dbm of noise is equivalent to the following readings:

8 dbrn = 6 dbrnc = 3 dba (1 kHz) = 0 dba (0-3 kHz)

c. Calculations of Noise Levels.

Example: Assume a measured noise level of -40 dbm. What is the level in terms of dbrn?

The level of -40 dbm is 50 db above the reference of -90 dbm (0 dbrn). The test instrument should therefore indicate a noise level of +50 dbrn.

d. Calculations of Relative Noise Levels.

Example 1: If the normal signal level is -3 dbm at the TLP at the time the noise level of +50 dbrn is being received, what is the relative level of the noise?
From the previous example, it is evident that +50 dbrn is equivalent to -40-dbm noise. Since -40 dbm is 37 db below the -3-dbm signal level at TLP, it is -37 dbm0 with the stated TLP.

Example 2: Is there another way to state relative noise levels?
Yes, there is. Since noise level is -90 dbm at 0 dbrn, relative value of noise level can also be stated with reference to the TLP. For example, the difference between 90 and 37 (-37 dbm0 in example 1 above) is -53 dbrn0 where the "0" after dbrn signifies noise level relative to the stated TLP.

e. Calculations of S/N.

Example 1: What is the S/N for the circuit discussed in b above?
The S/N is the difference between the desired signal level and the noise level. The relative value of -37 dbm0 therefore expresses the S/N as long as the signal level remains -3 dbm.
The S/N in this case is 37 db.

Example 2: If the input level drops from a normal of -3 dbm at the TLP to -18 dbm, what effect does this have on the previous S/N of 37 db?
The noise level of -40 dbm is now only 22 db below the signal level, so the S/N is 22 db. It is important to note, however, that if the circuit requirement is -37 dbm0, it remains at that value, because the relative value is taken with respect to the TLP (-3 dbm), not the actual signal level (-18 dbm) at the time. To meet circuit requirements of -37 dbm0, the noise will therefore have to be 37 db below -18 dbm, or -55 dbm.

2-6. THE MILITARY TRANSMISSION PLAN

The Military Transmission Plan provides an orderly method for allocating loss to the various type circuits found in any telephone communications network, and insures that a person anywhere in the network can converse satisfactorily with any other person in the network. The circuit conditioner is not responsible for the layout of the network, but he is responsible for conditioning designated lines within the network. He therefore must be aware of the important features of the plan.

a. Basic Rules. Conversation between two telephones is satisfactory only if the speech is loud enough to be heard and understood. With the military telephones currently in use, the maximum loss of the overall circuit between two telephones should not exceed 36 db. The total loss must be divided among the different types of circuits that connect the two telephones.

b. Types of Circuits. Types of circuits in a military telephone network are illustrated in figure 2-5. Voice power of the speaker at X in local area network 1 will be dissipated by the amount of loss in all circuits connecting with listener Y in local area network 3. By tracing the call from station X to station Y, you can see that the total loss is the sum of all losses distributed throughout the network. Tracing the call discloses the inclusion of the following network sections.

(1) Loops to the two tributary switchboards (SWBD).

(2) Two tributary trunks.

(3) Two tributary switchboards and three long-distance (LD) switchboards.

(4) Two long-distance four-wire trunks between LD switchboards.

(5) Supplemental LD trunks can be used for alternate routing if necessary. The supplemental trunk bypasses one LD switchboard.

c. Maximum Allowable Loss. Although maximum permissible loss from one telephone to another must not exceed 36 db, the preferred maximum is 20 db. The losses may be distributed in a number of ways. As a general rule, no more than two tributary trunks should be switched in tandem. Limitations of losses in a telephone system are as follows:

(1) Loops connected to a switchboard must not exceed 18 db.

(2) Loop plus tributary trunks must not exceed 9 db.

(3) LD trunk net loss must not exceed 6 db, and nominally should be 3 db.

(4) If maximum loss of an LD trunk is greater than 6 db but less than 12 db, it is called terminal grade. Terminal grade trunks may be connected to telephone loops, switchboard, or to tributary trunks only. They may not be connected to other LD grade facilities.

d. Long-Distance Trunks. Long-distance trunks extend between switching centers which may be separated by many hundreds of miles. Since losses over these great distances probably will exceed the limitation placed on LD trunks, amplifying equipment (repeaters) become essential. Such amplifying equipment raises the level of noise as it raises the level of signal. The transmission level diagram in figure 2-6 shows three repeaters in an LD trunk. The combination provides 3-db net loss; that is, 0 dbm input to the trunk results in -3-dbm output. To say it another way, receive level equals the send level minus the net loss. If the amplifying repeaters were not used, the total loss would be at least 45 db (25 db + 23 db), thus exceeding the maximum permissible loss.

(1) Input signal level of 0 dbm is raised to +6 dbm by 6-db gain in repeater 1.

(2) Line loss of 22 db drops the signal level to -16 dbm at the repeater 2 input.

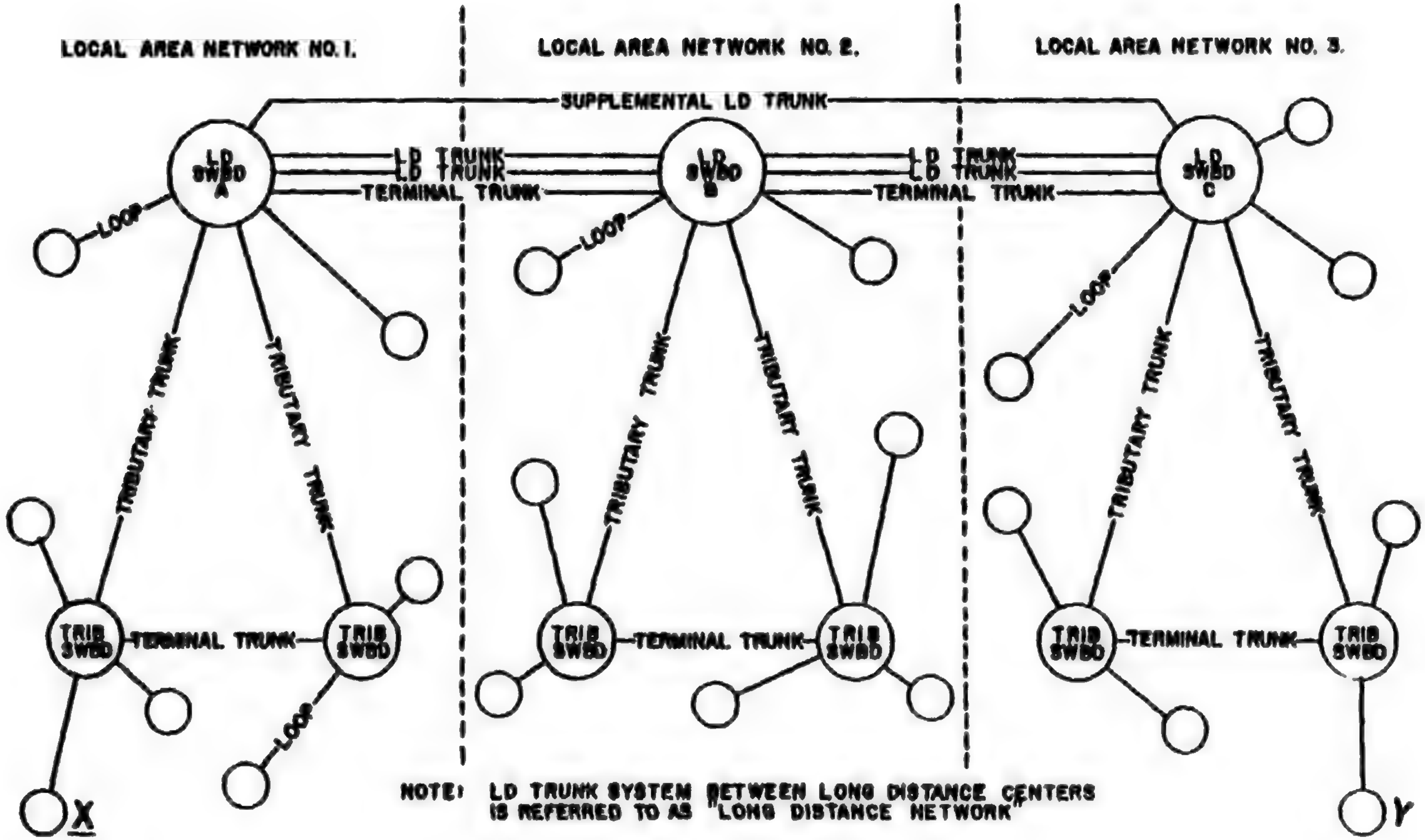


Figure 2-5. Types of circuits in a telephone network.

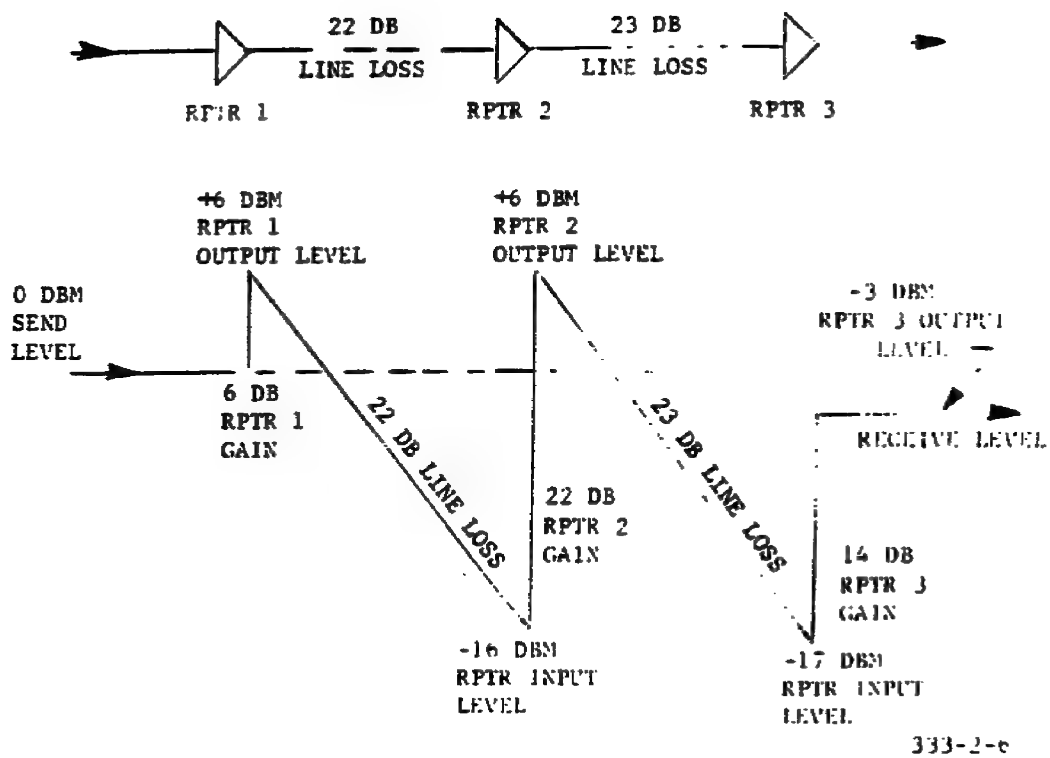


Figure 2-6. Transmission level diagram of LD trunk.

(3) Repeater 2 has 22-db gain to raise the signal level to +6 dbm.

(4) Line loss of 23 db drops the signal level to -17 dbm at the input to repeater 3.

(5) Gain of 14 db in repeater 3 raises the signal level to -3 dbm.

(6) Net loss is the difference between input and output levels, in this case 3 db.

STUDY EXERCISES

In each of the following exercises, select the ONE answer that BEST completes the statement or answers the question. Indicate your solution by circling the letter opposite the correct answer in the subcourse booklet.

1. Assume that the maximum gain of an amplifier is 35 db. If the overall amplifier-attenuator gain needed is 27 db, the technician must adjust the input attenuator to a loss of

a. 0 db.

b. 8 db.

c. 27 db.

d. 35 db.

2. When the circuit conditioner examines a frequency response curve for cutoff he looks for the point where the level drops 3 db from that at the midpoint of the bandpass. If the power level at midpoint of the bandpass is 8 mw, the scale in figure 2-4 shows that the cutoff is at a level of

a. 2 mw.

b. 4 mw.

c. 5 mw.

d. 11 mw.

3. Several factors bear on the reliability of voltage readings used to determine signal power levels. However, the most important combination of factors is the

a. value of measured voltage and circuit impedance.

b. circuit impedance and the frequency of the test signal.

c. accuracy of the instrument and the value of measured voltage.

d. frequency of the test signal and the accuracy of the instrument.

4. Assume that a 1,000-Hz test tone from the signal generator is applied to the ATT IN jack, and the level of -4 dbm is measured at the PAD OUT jack with the DB meter. If the voltmeter is used to measure the same level, table II in paragraph 2-2b shows that the meter should read

a. 1.23 volts.

b. 0.775 volt.

c. 0.489 volt.

d. 0.388 volt.

5. An accurate ac voltmeter may be used to establish a reference level of 0 dbm by taking the voltage drop across a calibrated resistor. The 1-mw standard will be obtained when the meter reads

a. 0.775 volt across a 600-ohm resistor.

b. 0.3873 volt across a 75-ohm resistor.

c. 0.2739 volt across a 150-ohm resistor.

d. 0.3073 volt across a 600-ohm resistor.

6. Assume that you measure the composite line signal from a four-channel FDM terminal, and find it to be +3 dbm. If each channel produces the same level, the power in each channel signal is

a. 1/2 mw.

b. 3/4 mw.

c. 1 mw.

d. 2 mw.

SITUATION

The schematic of one type of line terminating circuit is shown in figure 2-7. It connects the incoming line to station equipment. The circuit includes jacks, a 0-50-db variable attenuator, an equalizer having 10-db loss, a 42-db fixed-gain amplifier and a pad with a 6-db loss. Maximum undistorted output power capability of the amplifier is +34 dbm. Jacks permit inserting the signal generator, decibel, and voltmeter at various points in the circuit for testing. The input and output impedances of all circuit components are 600 ohms.

As a circuit conditioner, you will be expected to determine transmission losses and gains together with absolute and relative signal and noise levels. Further, you will be expected to make circuit adjustments to meet network requirements. The decibel and its derivatives are your tools.

Exercises 7 through 12 are based on this situation.

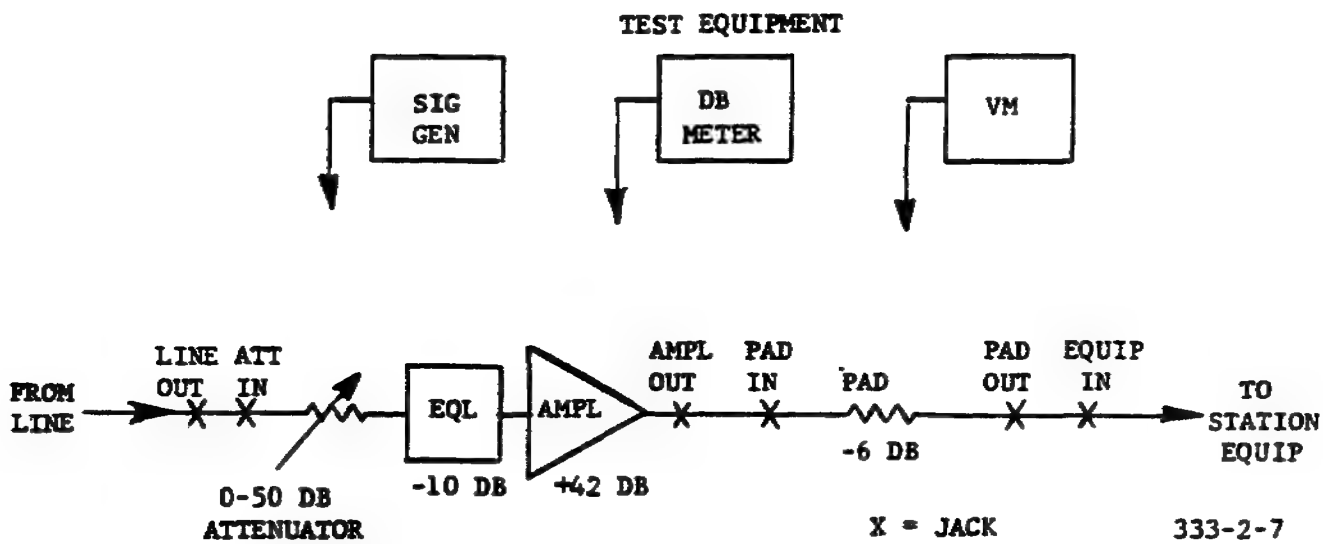


Figure 2-7. Schematic diagram of a line terminating circuit.

7. Assume that the face of the -decibel meter reads: 1 mw in 600 ohms = 0. This indicates that the decibel meter reads

a. gain of the amplifier.

b. loss of the circuit.

c. level in milliwatts.

d. level in dbm.

8. Assume that the circuit shown in figure 2-7 is carrying a frequency-division-multiplexed signal. If intermodulation distortion appears in the receiving channel outputs of the multiplex terminal, the most probable reason is that the

a. pad has changed value.

b. attenuator loss is excessive.

c. amplifier has been overloaded.

d. equalizer is incorrectly adjusted.

9. With a -12 dbm input level from the line and with an attenuator setting of 20 db, the expected level from the circuit to the equipment is

a. -2 dbm.

b. -4 dbm.

c. -6 dbm.

d. -8 dbm.

10. Assume that the TLP has been selected at AMP OUT jack with the level adjusted to -3 dbm. The relative level of the signal that is applied to the station equipment is

a. -6 db.

b. -9 dbm.

c. -6 dbm0.

d. -9 dbm0.

11. The technical controller or circuit conditioner may vary the overall gain of the circuit and thereby determine the output level by adjusting the

a. pad.

b. amplifier.

c. equalizer.

d. attenuator.

12. The maximum undistorted power output of the amplifier is +30 dbm. If this amplifier is delivering its rated output level, the corresponding output power delivered to the pad is

a. 1 watt.

b. 2 watts.

c. 1,001 watt.

d. 2 milliwatts.

NOTE: Use figure 2-1.

SITUATION

Assume that you are given a partial transmission level diagram for a long-distance (LD) trunk (fig. 2-8). The circuit contains three repeaters, and the received noise level is -50 dbm. The TLP is designated at the repeater 3 output with a level of +3 dbm. You are to determine the missing values and to analyze the diagram.

Exercises 13 thru 20 are based on this situation.

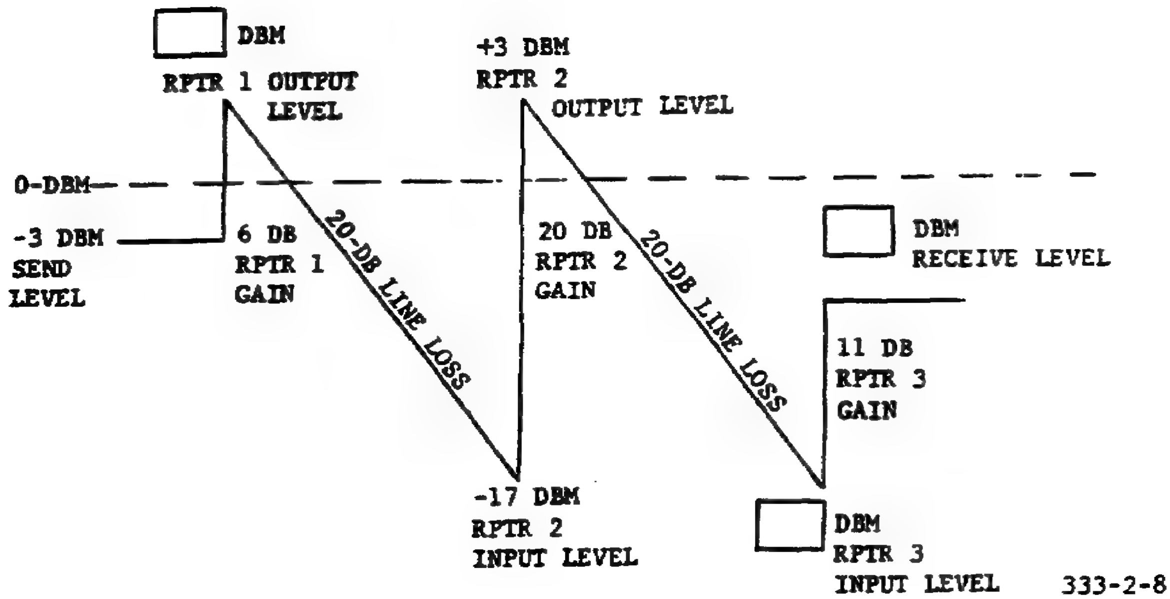


Figure 2-8. Partial transmission level diagram.

13. Approximately how many times will repeater 1 amplify the amount of power?

a. 2 times

b. 4 times

c. 6 times

d. 8 times

NOTE: See table I, paragraph 2-1.

14. The received noise level can be designated as

a. -20 dbrn.

b. -40 dbrn.

c. +40 dbrn.

d. +50 dbrn.

15. The noise level of -50 dbm can also be quoted relative to the level at the TLP (+3 dbm at the repeater 2 output). This noise level can be stated as

a. -33 dbm0.

b. -42 dbm0.

c. -53 dbm0.

d. -58 dbm0.

16. The technical controller mast be aware that any long-distance telephone communication system consists of several subsystems wherein each introduces its share of the total loss. The maximum loss permitted by the military transmission plan for any telephone network is

a. 3 db.

b. 6 db.

c. 18 db.

d. 36 db.

17. If the line has the nominal loss for an LD trunk, the receive level will be

a. -3 dbm.

b. -6 dbm.

c. -9 dbm.

d. -12 dbm.

18. Assume that the circuit conditioner finds that the net loss of the LD trunk rises to 9 db. He therefore advises the technical controller that the trunk is now limited to connection with

a. tributary trunks or other LD trunks.

b. other LD trunks or telephone loops.

c. a switchboard loop or other LD trunks.

d. telephone loops, switchboard loops, or tributary trunks.

19. What signal level can you expect to measure at the output of repeater 1?

a. +3 dbm

b. +6 dbm

c. +9 dbm

d. +12 dbm

20. What level can you expect to measure at the input to repeater 3?

a. -23 dbm

b. -17 dbm

c. -14 dbm

d. -11 dbm

CHECK YOUR ANSWERS WITH LESSON SOLUTION SHEET (PGS 61 and 62) AND MAKE NECESSARY CORRECTIONS.

LESSON 3

ECHO SUPPRESSORS AND COMPANDORS

SCOPE............................Echo suppression in long-lines transmission; use of compandors to minimize level variation and improve signal-to-noise ratio.

CREDIT HOURS.....................2

TEXT ASSIGNMENT..................Attached memorandum, para 3-1 thru 3-4; Appendix C, pages C-1 thru C-17

MATERIALS REQUIRED...............None

SUGGESTIONS......................Follow the study guide below.

LESSON OBJECTIVES

When you have completed this lesson, you should:

1. Know that the purpose of a compandor is to improve the signal-to-noise ratio for voice communication.

2. Know that a compandor amplifies weak and strong signals unequally.

3. Know that a compandor limits the variation in signal levels on the line.

4. Know that the presence of compandors and echo suppressors may be detrimental to data transmission over a telephone network.

5. Know that an echo suppressor controls the transmission path in a telephone network.

6. Know that unbalanced hybrid coil circuits may cause echoes in long-distance voice communication circuits.

7. Know that a hybrid coil circuit is necessary to convert two-wire to four-wire operation.

8. Be able to calculate signal levels passing through a hybrid coil circuit.

STUDY GUIDE

1. SOUND INTENSITY

Several arbitrary scales are used to describe sound levels in the electronic sciences. One type is used in Appendix C. The volume of sound (level) in this scale is based on the average threshold of hearing. The scale used identifies the threshold of hearing level as -60 db.

a. Whenever an arbitrary scale is indicated, you are advised to search for the reference level of that scale.

b. SSTS 53105 has used db's to indicate changes in power level as well as power level itself. The key to understanding the material in SSTS 53105 is to determine whether level or change of level is specified.

c. Because the arbitrary scale in SSTS 53105 does not agree with prior teaching in this subcourse, calculations of levels will not be required in the exercises.

2. SEQUENCE OF STUDY

a. Study the Attached Memorandum and answer exercises 1 through 10.

b. Study Appendix C and answer exercises 11 through 20.

ATTACHED MEMORANDUM

3-1. TELEPHONE NETWORKS

Telephone networks include all the telephones, switchboards, and trunks connecting the switchboards. Technical controllers and circuit conditioners must be able to think in terms of complete networks and all of the equipment within them. The total characteristic of a telephone network reflects the combination of characteristics of all individual items within the network.

a. Short-Distance Networks. Short-distance networks include telephones and switchboards together with wire pairs connecting the switchboards. One wire pair connects each telephone to the switchboard, and another connects the two switchboards together. As long as the overall circuit loss does not exceed the maximum permissible value of 36 db, the two-wire (pair) circuit will be satisfactory.

b. Long-Distance Networks. Long-distance networks include the usual telephones and switchboards found in short-distance networks. However, the over-all losses in long-distance networks may be so great that repeaters will have to be inserted in the wire or cable pair to raise the level. Further, the system may include telephone carrier systems and radio sets. Since repeaters and telephone carrier or radio systems use amplifiers for each direction of transmission, it is necessary to provide a pair of wires for each direction of transmission, thus making a four-wire (two-pair) circuit. A hybrid coil circuit will be necessary at each point of conversion from two-wire to four-wire operation. Each hybrid coil introduces losses which will be overcome by amplifiers in repeaters, or amplifiers in either carrier terminals or radio sets.

3-2. HYBRID COIL CIRCUIT

A hybrid coil serves as an "electronic traffic cop" that directs traffic through repeaters, as shown in figure 3-1. If the hybrid coil were not used, the output of one repeating amplifier would feed the input of the second, fording a closed amplifying loop which would "sing" or "howl." When the

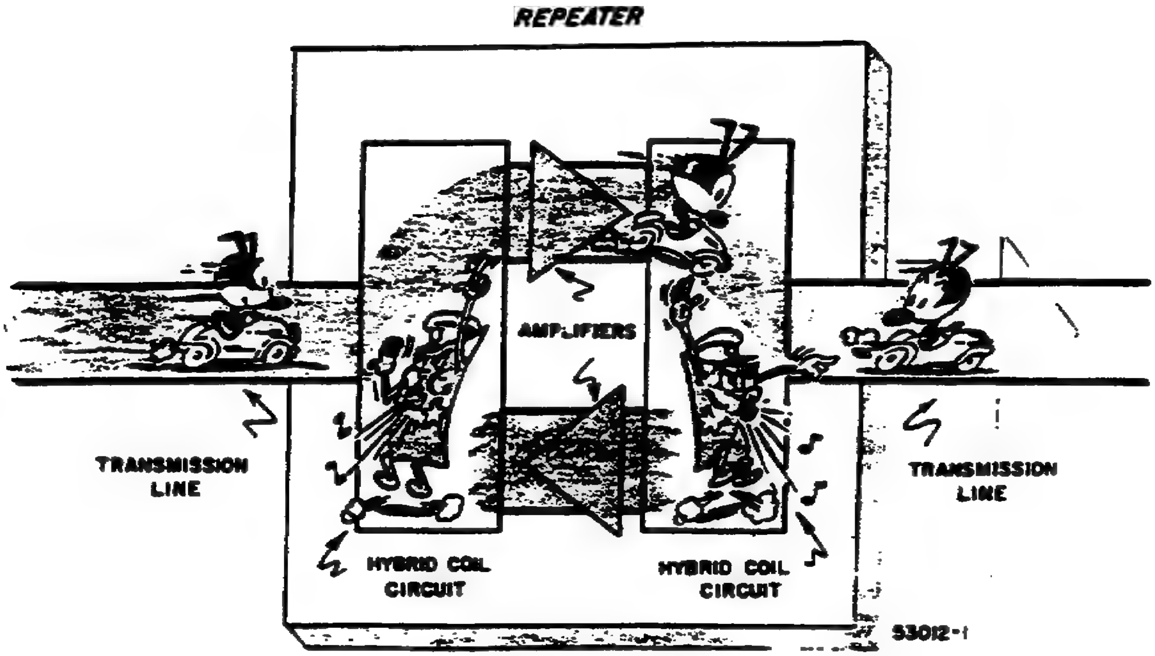


Figure 3-1. Hybrid coil directs traffic through repeaters.

hybrid circuits are adjusted correctly, voice energy arriving from the left transmission line travels through the top amplifier and out the transmission line to the right. Likewise, voice energy arriving from the right transmission line travels through the lower amplifier and out the transmission line to the left. Incorrect adjustment of hybrid circuits, especially the balancing networks, allows undesirable feedthrough from the upper amplifier to the lower, resulting in circuit singing. A hybrid coil circuit is essential in a two- to four-wire converter.

a. Schematic Diagram. The schematic diagram of a hybrid coil circuit illustrated in figure 3-2 shows at least five windings that are magnetically coupled because they are wound on the same transformer core. A balancing network represented by a variable resistor is used to separate talk (send) and listen (receive) paths.

b. Principles of Operation. One side of the hybrid circuit connects to a two-wire line, and the other side connects to a four-wire line. The two-wire line serves the switchboard, and the four-wire line serves the circuits between the switchboards including apparatus such as an echo suppressor or repeater amplifier. Each subscriber can talk and listen over the two-wire line. However, the local subscriber's voice travels to the distant station on one pair of the four-wire line, and the distant subscriber's voice returns through the second pair of the four-wire line.

(1) A hybrid circuit is essentially a balanced bridge. To operate properly, the balancing network must be electrically equal to the impedance of the switchboard loop, and the impedance of the send line must equal that of the receive line. An imbalance of either of the two lines will cause feedthrough as shown by the dashed line. Another cause for feedthrough is the wrong value of a balancing resistor.

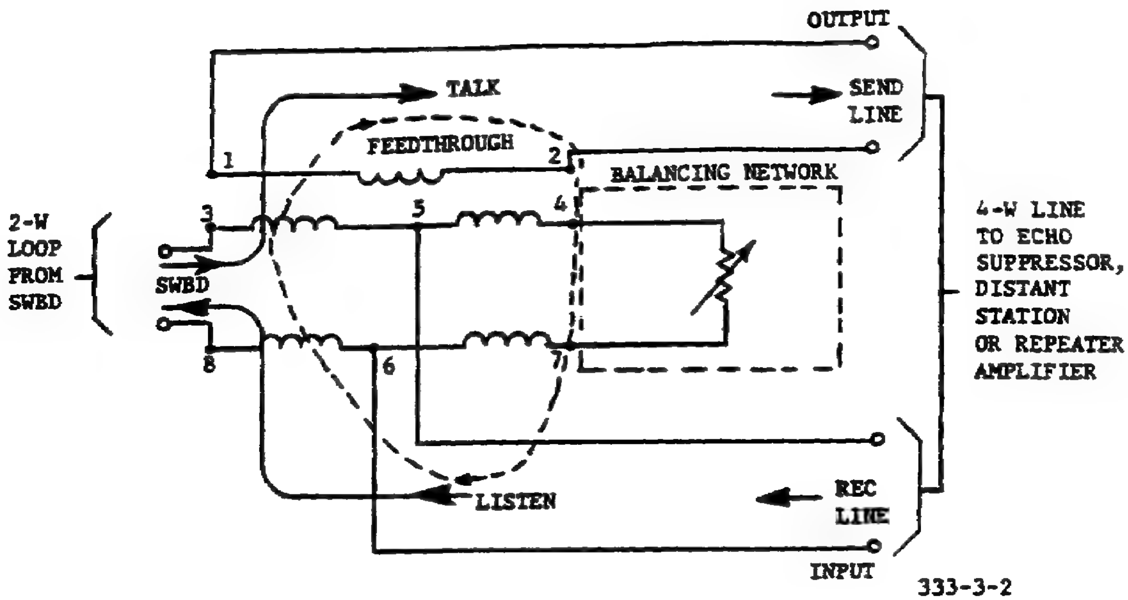


Figure 3-2. Hybrid coil circuit, schematic diagram.

(2) Since at least half of the energy passing through the hybrid circuit in either direction of transmission is dissipated in the balancing network, there will be at least a 3-db loss in either the talk or listen signal path. If the loss is not acceptable, an amplifier must be inserted in the line to raise the signal level an amount equal to the loss.

(3) The normal loss through a hybrid coil circuit is 4 db, meaning that a signal of 0 dbm from the switchboard will leave the send line at a level of approximately -4 dbm. Likewise, a signal of 0 dbm arriving at the input to the receive line will arrive at the switchboard with a level of approximately -4 dbm.

c. Delay. Because of the large lumped inductance in the windings of a hybrid coil, envelope delay may be excessive during transmission of high-speed data signals. The usual practice in data transmission is to use four-wire operation to eliminate possible delay caused by hybrid coils.

3-3. ECHO SUPPRESSOR

a. Cause for Echo. In repeaters or carrier terminal sets, an amplifier is used to raise the voice level in each direction of transmission. If the feedthrough from the receive line to the send line is great enough, the circuit may sing. If the feedback is not sufficient to make the circuit sing, the talker may be able to hear his own voice come back to him like an echo. In short lines, the echo will have no effect because the time lapse is small, but if the lines are long, the delay of the signal as it returns to the talker may cause the echo to be noticeable. An echo that is loud and delayed may cause the talker to experience serious difficulty conversing over the circuit.

(1) Hopefully, the circuits connected to the hybrid coil will balance correctly. No singing will be noticed and no echo will appear. Since this is rarely the case in practice, long-distance telephone networks are often provided with echo suppressors.

(2) Some of the major causes of unbalance of hybrid coils include wrong values of balancing resistors, temperature change of the wires or cables, leakage, variation in line matching impedance, and reflection. Secondary causes include changes in traffic routing, improperly adjusted amplifiers, and changes in types of terminal equipment.

(3) A long-distance telephone system may exhibit many echoes, one echo resulting from each impedance mismatch or unbalanced hybrid. Echoes from the greatest distance are usually more noticeable because of greatest time delay, but fortunately they are usually weakest since the line loss is greatest between the echo source to the talker.

b. Echo Suppressor Principles. The simplest form of echo suppressor is shown in figure 3-3. It is essentially a voice-controlled circuit wherein the voice passing through the circuit locks up the remaining path.

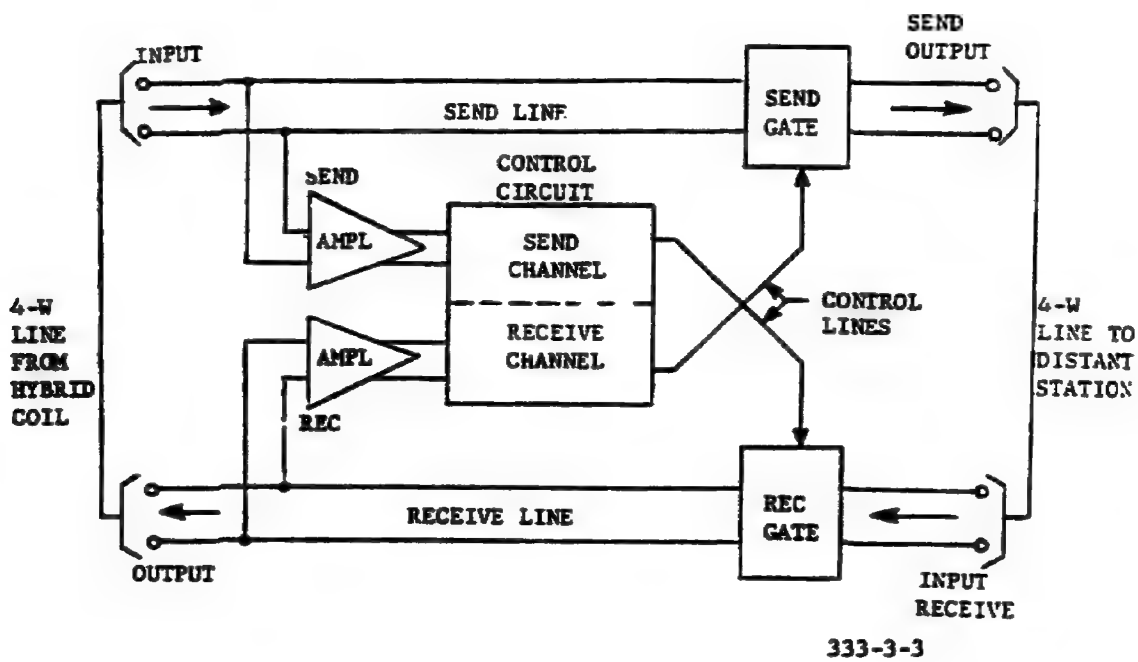


Figure 3-3. Echo suppressor circuit, block diagram.

(1) One side of the echo suppressor circuit connects to the hybrid coil by a four-wire line and the other side terminates in a four-wire line to the distant station. Each direction of transmission has its own amplifier to raise the level to the value needed to operate the control circuit. Control lines cross, so that the send channel controls the receive gate and the receive channel controls the send gate.

(2) When the voice of the first talker passes over the send line the amplified signal enters the send channel of the control circuit to develop a control voltage for closing of the receive gate. As long as the talker continues to speak, the receive line is locked up and no signal is received by him. When he stops talking, the send channel opens the receive gate. If a voice signal enters the receive line from the distant station, a sample of the signal is amplified for use by the control circuit. The receive channel closes the send gate, thus preventing a signal from passing in the talk direction. Only one person at a time can talk on such a circuit, and the talker cannot hear echoes of his own voice.

(3) Since the activating circuits require a short period of time for operation, the first impulse of signal will be lost. This can be detrimental to data signals. For this reason, the usual practice is to switch the echo suppressor out of the telephone network during data transmission.

c. Trunk Circuits. One effective method for minimizing the development of echo without resorting to echo suppressors is to use four-wire trunks. This change eliminates the hybrid coils at repeaters and, with them, lessens the need for coil and line balance. Level compensation becomes simpler and circuit patching is accomplished with less difficulty.

3-4. COMPANDOR

To be useful in telephone communication, a compandor must have a path for each direction of transmission, as shown in figure 3-4. Four-wire operation is needed to establish the two independent directions of transmission. A hybrid coil is therefore necessary to convert the 2-W switchboard loop to 4-W operation on the line between terminals.

a. Voice Distortion. Distortion of voice is noticeable when the expandor and compressor units are improperly adjusted, giving rise to incompatible compression and expansion ratios. Distortion will also be noticed if attack and recovery times are unsynchronized. Quality of voice transmission may suffer, but still be understandable for all normal voice communication. Trouble develops, however, when signals other than normal voice are transmitted over the circuit containing compandors.

b. Restrictions. Compandors are beneficial when used with long-distance voice or facsimile (analog) communication, because of their ability to improve the signal-to-noise ratio. They are usually detrimental, however, when digitized signals are passed over the circuit. The time delay in attack and recovery circuits may exceed the tolerance for envelope delay. When data signals are transmitted over a voice facility containing compandors, the compandors should be switched out of the telephone network unless the technical characteristics indicate freedom from digital distortion.

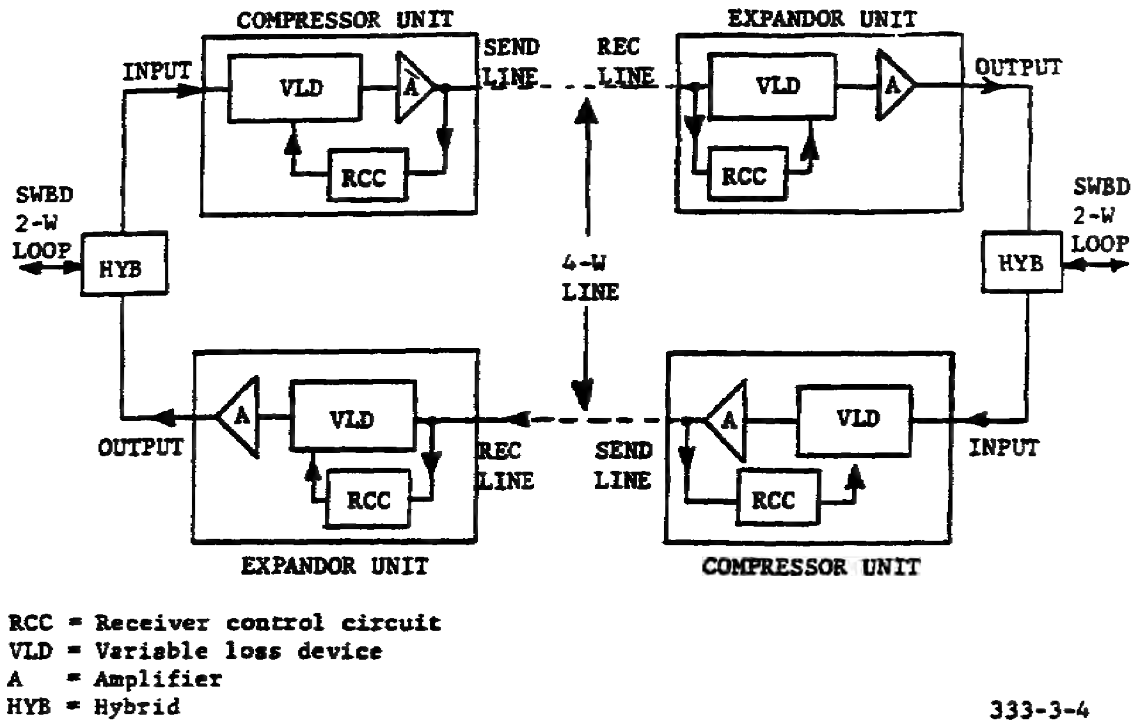


Figure 3-4. Compandor in a telephone communications circuit.

STUDY EXERCISES

In each of the following exercises, select the ONE answer that BEST completes the statement or answers the question. Indicate your solution by circling the letter opposite the correct answer in the subcourse booklet.

1. A complete telephone communication system my include short- and long-distance networks. One difference between these two types of networks is that the long-distance network includes

a. switchboards and repeaters.

b. telephones and switchboards.

c. repeaters and telephone carrier systems.

d. telephone carrier systems end switchboards.

2. At each point of conversion from two-wire to four-wire circuits, it is always necessary to insert a

a. switchboard.

b. patching panel.

c. repeating amplifier.

d. hybrid coil circuit.

3. The function of the balancing network of the hybrid coil circuit shown in figure 3-2 is to

a. minimize line loss.

b. balance amplifier gain.

c. prevent singing of circuits.

d. eliminate hysteresis of transformer core.

4. Selection of the correct value of resistance for the balancing resistor in a hybrid coil circuit is important because an incorrect value will cause

a. losses of signal energy.

b. feedthrough between talk and listen paths.

c. noise buildup through magnetic coupling of the coils.

d. transfer of signals from the switchboards to the lines.

5. The function of a hybrid coil circuit is similar to the function of a

a. balanced bridge.

b. phantom circuit.

c. repeating coil.

d. loading coil.

6. Assume that a level of -2 dbm enters a hybrid coil circuit on the two-wire line from the switchboard. If the hybrid circuit has normal loss, the level going out on the send line will be approximately

a. -1 dbm.

b. -2 dbm.

c. -4 dbm.

d. -6 dbm.

7. One disadvantage of a hybrid coil in a telephone network carrying data signals is that the hybrid circuit may introduce excessive

a. loss.

b. delay.

c. resistance.

d. feedthrough.

8. Assume that a telephone user hears a number of echoes on a long-distance telephone network. The echo that has traveled the longest distance causes the most interference because this echo is

a. delayed the longest.

b. louder than the others.

c. unaffected by line loss.

d. reinforced by the other echoes.

9. Assume that a long-distance telephone network uses echo suppressors. An echo will not be heard by the talker as long as he speaks because the

a. send channel controls the send gate.

b. send channel controls the receive gate.

c. receive channel controls the send gate.

d. receive channel controls the receive gate.

10. Assume that echo suppressors are not usable in a long-distance telephone network passing data signals. One effective method for passing the traffic without developing echoes is to use

a. hybrid coils.

b. two-wire trunks.

c. four-wire trunks.

d. repeating amplifiers.

11. Host telephones are designed to transmit the bandwidth from approximately 200 to 3,200 Hz because

a. compandors work most effectively in this range.

b. noise and crosstalk have the least effect in this range.

c. all the vocal tones in a human voice are included in this range.

d. most of the energy of speech signals is concentrated in this range.

12. The overall action of a compressor during telephone transmission is to

a. attenuate weak and strong signals equally.

b. raise the power level of both weak and strong signals equally.

c. raise the power level of weak signals and attenuate strong signals.

d. attenuate the weak signals and raise the power level of strong signals.

13. A compandor is useful in improving the signal-to-noise ratio in a telephone communication system. It is called a remedial device because it

a. gives relief from the effects of noise on weak signals.

b. eliminates any noise and crosstalk from the incoming signal.

c. filters out the noise above and below the voice-frequency range.

d. shifts the frequency of voice components which lie beyond the voice frequency range.

14. A volume compandor reacts more slowly than an instantaneous compandor because, unlike the instantaneous compandor, the volume compandor is designed to compensate for

a. intensity range.

b. individual amplitude peaks.

c. amplitude of syllabic variation.

d. frequency of syllabic variation.

15. Data communication systems use pulses similar to pulse-code modulation (PCM). This means that when data are transmitted, the type of compandor used must be the

a. volume.

b. syllabic.

c. modulated.

d. instantaneous.

16. The three basic parts of a volume compandor are

a. amplifier, peak clipper, and variable loss device.

b. rectifier control circuit, amplifier, and peak clipper.

a. variable loss device, rectifier control circuit, and amplifier.

d. peak clipper, variable loss device, and rectifier control circuit.

17. In the expandor unit of a compandor the control current varies the amount of attenuation. What are the relationships among the signal, control current, and attenuation?

a. A weak signal produces a large control current, and attenuation is high.

b. A weak signal produces a small control current, and attenuation is low.

c. A strong signal produces a large control current, and attenuation is low.

d. A strong signal produces a small control current, and attenuation is high.

18. Assume that the compression ratio in a compandor is 2:1 and the conmpanding range is 50 db. The intensity range entering the expandor is

a. 100 db.

b. 75 db.

c. 50 db.

d. 25 db.

19. Assume that the compression ratio of a compressor is 2:1. This means that in the compressor the

a. intensity range of the line signal is one-half the intensity range of the input signal.

b. power transmitted between compandors is one-half the input value.

c. loss between compandors is one-half the value that would exist without compandors.

d. level changes to one-half its former value when passing through the compressor.

20. One characteristic of a syllabic compandor is that it has

a. zero attack time.

b. zero recovery time.

c. equal attack and recovery times.

d. a longer recovery than attack time.

CHECK YOUR ANSWER WITH LESSON SOLUTION SHEETS, PAGES 63 AND 64 AND MAKE NECESSARY CORRECTIONS.

LESSON 4

EQUALIZATION

SCOPE............................Equalizing telephone circuits to improve amplitude-frequency relationships and to minimize envelope delay.

CREDIT HOURS.....................1

TEXT ASSIGNMENT..................Attached Memorandum, para 4-1 thru 4-6

MATERIALS REQUIRED...............None

SUGGESTIONS......................None

LESSON OBJECTIVES

When you have completed this lesson, you should:

1. Be aware of the importance of equalization of a telephone line.

2. Know that amplitude-frequency distortion of a telephone line affect voice quality and may also affect data signals.

3. Know that envelope delay causes distortion of data signals, but has little effect on voice quality.

4. Know that equalization inserts corrective networks to compensate for line characteristics.

5. Know that equalization is always accomplished at the receiving end of a line.

6. Be able to analyze characteristics of circuits presented on graphs of amplitude-frequency and envelope-delay response curves.

7. Be able to determine delay values needed to compensate for envelope delay.

8. Be able to determine amplifier gain to compensate for line and equalizer loss.

ATTACHED MEMORANDUM

4-1. IMPORTANCE OF EQUALIZATION

All operations involving circuit conditioning use equalizers. These units are designed to equalize the amplitude-frequency response of a telephone circuit. Best quality of voice or sound reproduction is achieved when the response of a circuit is constant over the entire spectrum of frequencies being

used. Equalizers give technicians tools to correct for distortions that communication circuits introduce into the telephone signals.

a. Analog Signal Distortions. A voice signal is a form of analog signal. Analog signals vary in amplitude and frequency, and theoretically have no discontinuity (break) in the signal waveform.

(1) Harmonic distortion. Harmonic distortion is minimized by the assurance of correct operation of amplifiers and other circuit constants within their amplitude capabilities. In other words, technicians must make every effort to see that repeaters and amplifiers in carrier terminals are not overloaded. Equalizers can do nothing to minimize harmonic distortion once it is produced. Harmonic distortion produces additional frequencies not present in the original signal.

(2) Frequency distortion. Frequency distortion of an analog signal is produced mostly in equipment items which modulate a carrier with sound energy. Frequency distortion is minimized by the assurance of identical carrier frequencies for modulation and demodulation wherever possible. Equalizers cannot correct for frequency distortion. Frequency distortion causes a shift in all the frequencies in a received signal simultaneously.

(3) Amplitude-frequency distortion. When an analog voice signal leaves the output of a terminating device, such as a telephone or a line amplifier, there is a relationship between the amplitude and frequency of the signal, which should be maintained throughout the circuit. Any variation in the received signal between the amplitude-frequency relationship is amplitude-frequency distortion. This is the type of distortion that line equalizers were originally designed to minimize.

b. Digital Signal Distortions. Most digital communication circuits use modems; these are combination modulators and demodulators. Digital signals at the sending end enter the modulator as dc pulses and leave the demodulator as dc pulses. The line signals between the modems usually carry tone signals. One task of the technician is to maintain the quality of these line signals throughout the length of the circuit. The intelligence within the digital signals is carried in the transitions. Every effort is made to minimize the effect of the circuit on the received signal transitions, and thus minimize the effect of distortion. It is therefore desirable that all received frequencies in the signal maintain the same amplitude, frequency, and phase relationships- as when transmitted.

(1) Amplitude distortion. The voice-frequency (VF) signals that carry the intelligence contain enough of the transitional frequencies to allow the demodulator to correctly reproduce the digital pulses, with a minimum of error and distortion. Maintaining the amplitude of these frequencies is not a difficult technical matter. Line amplifiers overcome line loss. Finally, a combination of amplifiers, limiters, and rectifiers in the demodulator reproduces the digital pulses to their original form.

(2) Frequency distortion. Frequency distortion is always a possibility in circuits containing reactive elements (capacitance, inductance). Distortions result when transitional frequencies are displaced in phase. Further, the higher order harmonics in the transitions are attenuated more than the lower order harmonics, giving the demodulated pulse a modified waveshape compared with the original.

(3) Phase delay or envelope delay. Phase, or envelope, delay is by far the most troublesome characteristic in a circuit intended to convey digital signals. Theoretically, all component frequencies that give a pulsed digital signal its characteristic waveshape are present in the transmitted signal in correct amplitude, frequency, and phase. When the digital signal modulates a carrier for transmission over a circuit, all the component frequencies should retain their amplitude, frequency, and phase relationships. However, reactive elements within the circuits cause frequency distortion as well as phase distortion of the high-order harmonics in the signal transitions. Nonlinear phase shift of these frequency components delays recognition time of the demodulated data pulses. As long as the phase shift is directly proportional to the change in frequency, the delay of the demodulated data pulses remains constant. Nonlinearity of phase shift (envelope delay) with respect to change of frequency is corrected by an equalizer designed to make all envelope delays equal to the longest (maximum) delay of any received signal element. There is no way to shorten delays that already exist in the envelopes of the received signal waveforms.

(4) Pulse delay, or delay distortion. Envelope delay of the circuit affects pulse recognition delay in demodulators. When properly equalized, the circuit has equal-length delays. Under this condition, the demodulated digital pulses are all delayed an equal amount. Since all pulses are similarly affected, delay distortion of pulses in the signal is minimized. Envelope delay and pulse delay are therefore related functions. However, each of these two items is measured in a different manner. Since envelope delay varies with frequency in an unpredictable manner in an uncorrected circuit, the effect of delay is measured at selected frequencies. On the other hand, pulse delay is measured in percentage of pulse-length variation. However, the technician can quickly identify the presence of envelope delay by observing the waveshape of demodulated pulses, and measuring their lengths. When received pulses exhibit their original waveforms, along with their normal lengths, the technician can be assured of two characteristics: envelope delay of the circuit is proportional to frequency, and all pulses in the signal are delayed an equal length of time. A properly equalized (corrected) circuit produces these desirable qualities.

(5) Summation. In summery, envelope delay and delay distortion are related terms, although measured by different instruments. Further, by making all envelope-delay values uniform by the equalizing process all digital pulses are uniformly delayed. The primary purpose of equalizing envelope delay of the voice circuit is to minimize the effect of delay distortion on received dc digital signal.

4-2. AMPLITUDE-FREQUENCY EQUALIZER

a. Characteristics. An ideal telephone pair is a "flat-loss line." That is, the loss is constant over all frequencies in the voice range. The graph of such a telephone pair is characteristically a horizontal line. In figure 4-1 it is labeled "desired flat loss." Unfortunately, a long telephone pair will seldom exhibit uniform loss characteristics. Normally, the loss increases with a rise in frequency as indicated on the line-loss curve.

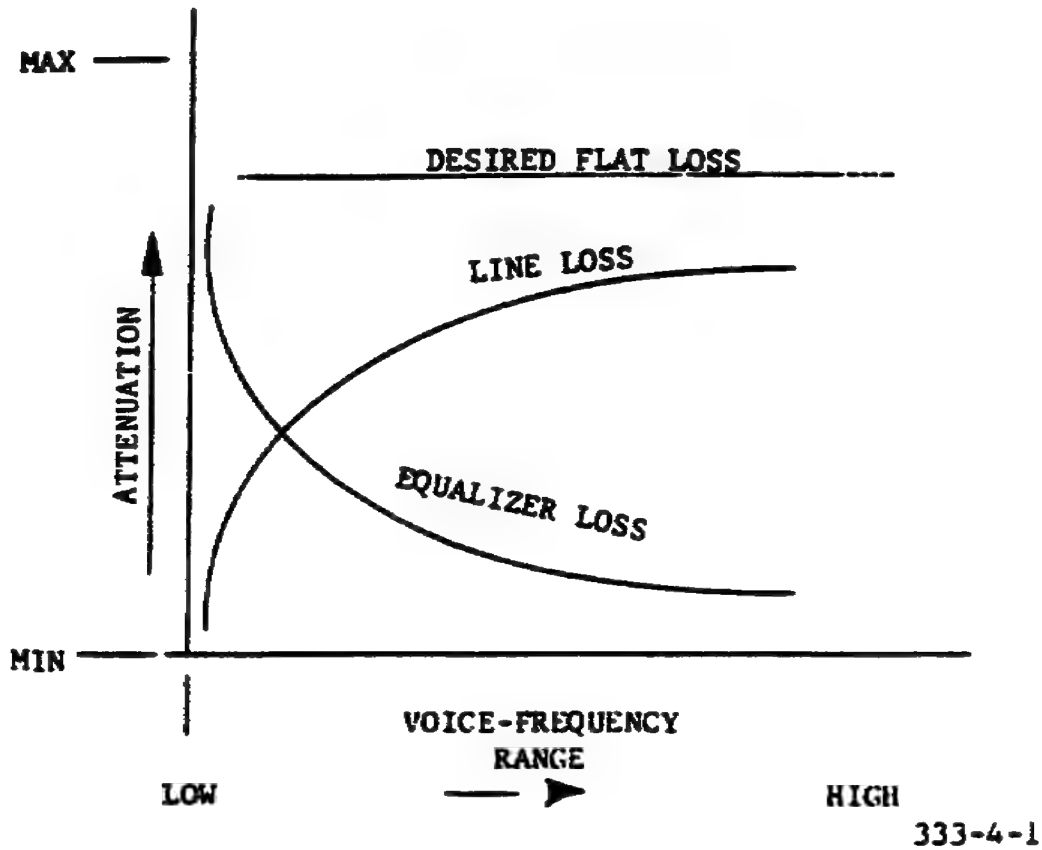


Figure 4-1. Characteristics of a voice-frequency equalizer.

(1) To obtain the desired flat loss, it is necessary to counteract the effect of the line constants which produce the line-loss curve. This is theoretically accomplished by inserting the equivalent lumped reactive constants which will produce the exact opposite equalizer-loss curve. If the mirror image is complete, the result is a flat-loss line.

(2) The desired flat loss is always greater than either the line or the equalizer loss, since the flat line loss results from the sum of both line and equalizer loss. An amplifier must either precede or follow the equalizer to compensate for the total loss.

(3) The amplifier used to raise the signal level also has its own gain characteristic. Combining the equalizer and amplifier may cause the characteristic to lack uniform response. The recommended procedure

to equalize for amplitude-frequency response calls for measuring the level at the output of the amplifier. In this way, the equalizing process will take into account line loss, equalizer loss, and amplifier gain characteristic.

<u>b</u>. <u>Circuit</u>. An amplitude-frequency equalizer circuit (fig. 4-2) contains two-basic circuits; one is for low-frequency, and the second is for high-frequency equalization. The balance between the two circuits in the adjustment process takes care of the frequencies which lie between the two limits; these are called midfrequencies.

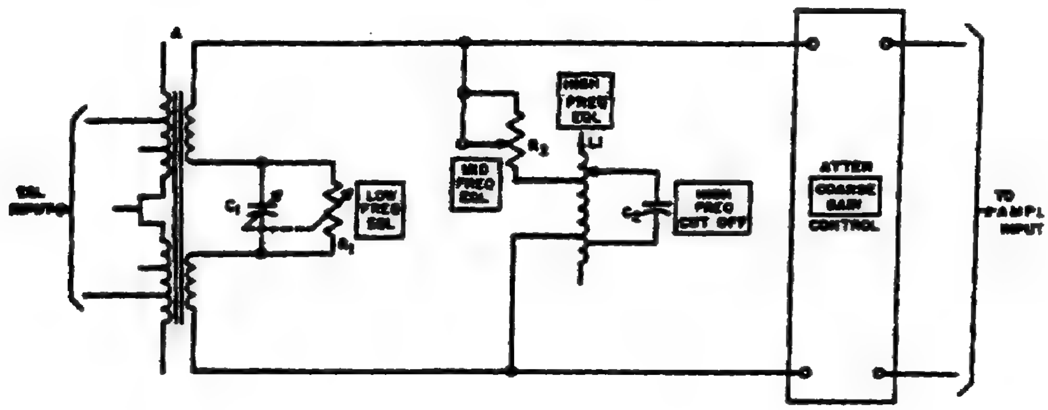


Figure 4-2. Amplitude-frequency equalizer, simplified schematic diagram.

(1) Essentially, the equalizer circuit consists of a series-tuned circuit (secondary winding of transformer A and capacitor C1) that can be tuned to some frequency below the lowest line frequency. The series-tuned circuit has little effect on the higher frequencies. Resistor R1 adjusts the circuit Q, thus broadening the response as needed. Further, the low impedance of the tuned circuit near resonance tends to pass the low frequencies more readily than high frequencies, thus causing greater loss (dissipation) in resistors R1 and R2.

(2) A parallel-tuned circuit consisting of L1 and C2 is adjusted to resonate at some frequency above the highest line frequency. The tuned circuit will therefore present highest impedance at the top frequencies, but has little effect at the low frequencies. Resistor R2 serves to lower the Q to the degree necessary to broaden the response curve. The overlap of the broadened curves of both tuned circuits determines the loss at midfrequencies.

(3) The combination of all adjustable elements in the equalizer produces a variable-loss bandpass filter having cutoff frequencies below the lowest frequency passed and above the highest frequency passed.

(4) The overall loss of the equalizer is unknown and must be compensated by gain of an amplifier. To minimize the development of harmonics, a fixed-gain amplifier usually follows the output circuit of the equalizer. It is therefore necessary to use an attenuator between the equalizer output and the amplifier input to adjust the output level of the amplifier to the proper value.

(5) Normally only one amplifier is needed to amplify the full range of voice frequencies. With this arrangement, the amplifier makes no change in the frequency characteristic or response of the line and equalizer; it simply raises the level of any frequencies present. Although the amplitude-frequency type of equalizer has the advantage of simplicity, levels of narrow frequency bands within the voice-frequency range cannot be adjusted because of the single amplifier used.

4-3. COMBINATION EQUALIZERS

A combination equalizer is designed for both amplitude-frequency and envelope-delay compensation. Circuit conditioners will use combination equalizers to condition telephone lines for data transmission.

a. Delay Characteristics. A graph of delay characteristics is shown in figure 4-3. The delays at different frequencies in the voice range of a selected telephone line are indicated by the uncorrected circuit curve. To minimize distortion of data signals passing through this telephone line, it is

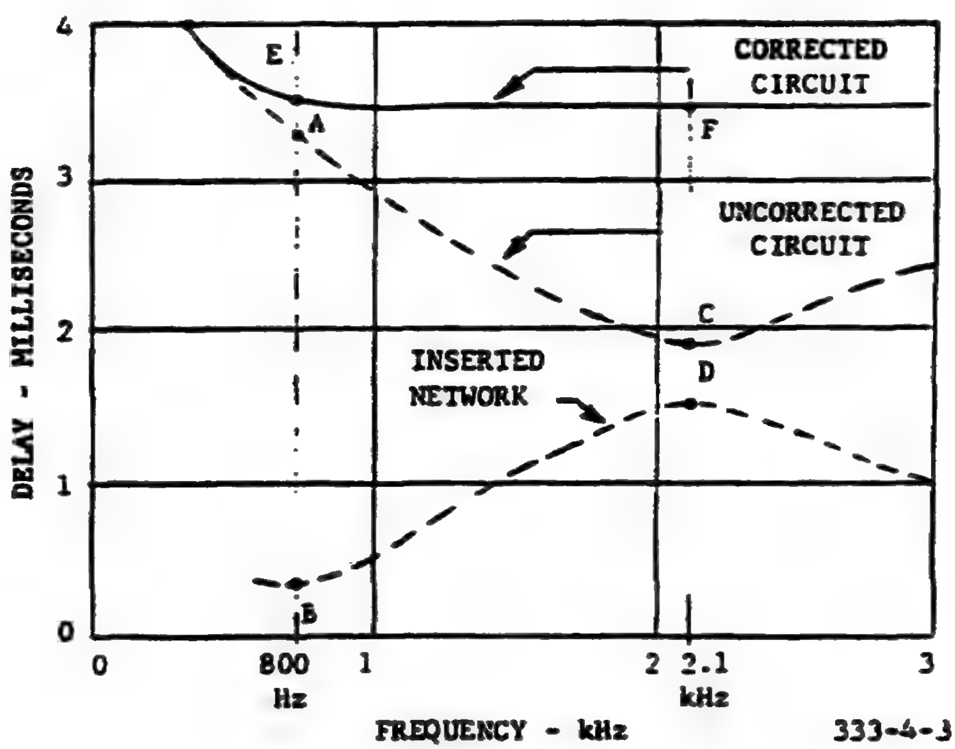


Figure 4-3. Delay characteristics of a combination equalizer.

desirable to have uniform delays across the voice-frequency band. This ideal condition is indicated by the corrected circuit curve. Notice that the "flat" characteristic here means uniform delays at each of the various test frequencies. Keep in mind that circuit parameters, stated in terms of envelope delay, relate delay time of the carrier modulation envelope to various test frequencies in the spectrum. In order to make all delays uniform, it is necessary to insert a network. The sum of delays of the line and inserted network is nearly constant. However, the difference between delays is also a constant, but is never zero. Another name for the inserted network is a delay equalizer.

(1) Uniform delay is achieved when the inserted network delay curve is an exact mirror image of the delay curve of the uncorrected circuit. In practice, this desirable condition is difficult to achieve, but the circuit conditioner should strive for it.

(2) Since variation of amplitude-frequency response and delay are inter-dependent, the circuit conditioner usually must compromise both conditions to give the best combination of characteristics.

(3) In practice, the technician does not measure delay in the inserted network (equalizer). His first operation is to plot the envelope-delay characteristics of the uncorrected circuit from readings taken at several selected frequencies. He then inspects the characteristic curve to determine whether parameters for that circuit are satisfied. If parameters are satisfied, the line or circuit needs no correction, and he needs no delay equalizing, even though amplitude-frequency equalizing may still be necessary. If delay equalization is required, he inserts estimated network segments in the circuit so that the total delays of circuit and network approach similar values. Further adjustment makes the delays at all selected frequencies nearly uniform. The ideal condition is reached when all delays are identical, in which case the characteristic becomes a straight line across the graph.

(4) The following examples are based on figure 4-3. Assume that the frequency range is from 800 Hz to 2.1 kHz, and that maximum delay is 3.5 milliseconds (ms).

<u>Example 1</u>. What is the delay of the uncorrected circuit at 1 kHz?

Solution: The uncorrected circuit curve intersects the 1-kHz frequency line at approximately 2.9 ms.

<u>Example 2</u>. How much delay is inserted at 1 kHz to provide the corrected circuit delay of 3.5 ms?

Solution: Inserted delay = corrected delay - uncorrected delay
= 3.5 - 2.9 = 0.6 ms.

<u>Example 3</u>. What are the delays at 800 Hz?

Solution: Uncorrected delay (point A) = 3.2 ms.
Inserted delay (point B) = 0.3 ms.
Corrected delay (point E) = 3.5 ms.

Example 4. What are the delays at 2.1 kHz?

Solution: Uncorrected delay (point D) = 1.6 ms.
Inserted delay (point C) = 1.9 ms.
Corrected delay (point F) = 3.5 ms.

b. Block Diagram. Figure 4-4 shows the block diagram of a typical combination equalizer that is designed to compensate for both amplitude-frequency distortion and envelope delay. It is similar to a simple amplitude-frequency equalizer, but it contains 12 equalizer sections, each capable of equalizing a small segment of the frequency spectrum in the voice range. Each section is tunable so that selected segments of the spectrum may be equalized independently. Not all sections need be used; the number required depends on the parameters specified for the service and the characteristics of the transmission facilities. For simple voice communication an equalizer may not be necessary, but if the circuit is to be switched for services requiring conditioning, the number of equalizer sections will be chosen to satisfy the highest type of service to be placed on that circuit. For data transmission an equalizer is essential, with most of the sections being used.

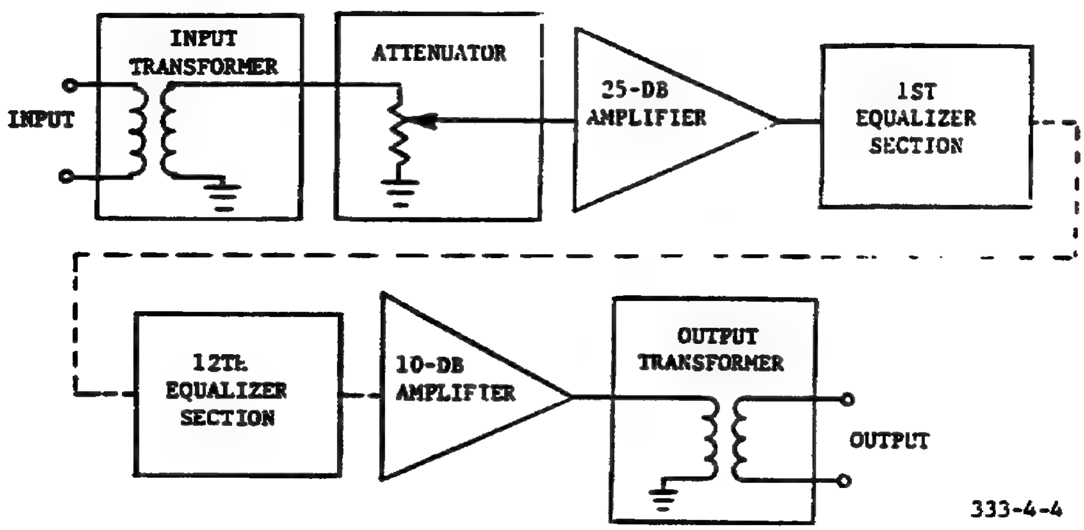


Figure 4-4. Combination equalizer, block diagram.

c. Equalizer Section. One of the 12 equalizer sections is shown in figure 4-5. The frequency can be adjusted to locate the point where equalization is most needed within the narrow spectrum selected by the section. Envelope delay can be adjusted to make all delays equal in the 12 sections (or number of sections used). Amplitude can be adjusted so that the output level of each section has an approximately equal value. The circuit conditioner will find that variation in any of the three controls (frequency, delay, or amplitude) has an

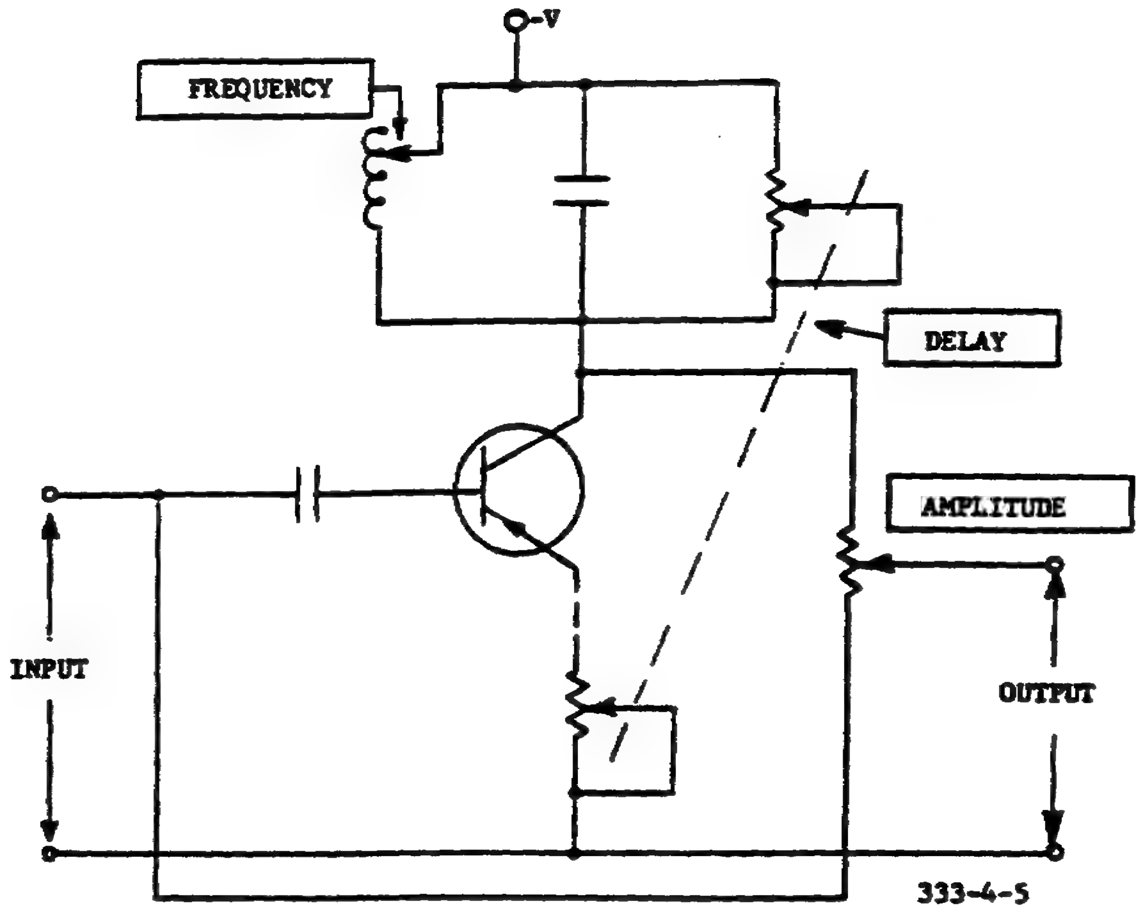


Figure 4-5. One equalizer section, schematic diagram.

effect on the other, so he attempts to strike the best compromise attainable. Note that each equalizer section contains its own amplifier, thus each section can be operated independently. The line amplifiers that are common to all spectrum segments amplify all of them together to the extent necessary to compensate for line and equalizer loss. Six of the sections have a frequency range of 0.5 to 2.2 kHz, and six have a range of 1.2 to 3.5 kHz. With this arrangement, there is an overlap of 1.2 to 2.2 kHz to assure that equalization is available anywhere within the voice band of 0.5 to 3.5 kHz. In the equalizing process the circuit conditioner uses the number of sections he finds necessary to satisfy requirements. During the adjustment procedure he adjusts each of the sections used to a different frequency segment in the spectrum designated by the parameters.

4-4. EQUALIZATION PROCEDURE

Equipment now in use is equalized empirically; that is, known signal frequencies at uniform levels are transmitted from one end of the system. At the opposite, or receiving, end of the system, the circuit conditioner adjusts the controls of the equalizer while watching meter readings. This procedure is time consuming, and requires a great deal of cooperation and patience on the

part of both technicians. Normally the first adjustment is for amplitude-frequency response, followed by envelope delay. Separate test sets are used to measure the two different parameters (amplitude-frequency response and envelope delay). Subsequent adjustments are made to minimize variation in the parameters. This is made necessary by the changes that take place in one parameter while an adjustment of the equalizer is made to satisfy another parameter. Knowledge of the system is important to the procedure. The circuit conditioner muse be given a circuit layout card so that every item of equipment is known to him. Equipment characteristics, especially frequency cutoff, are important because no amount of conditioning can compensate beyond that response limitation. A normal measurement of system parameters cannot disclose characteristics of equipment used in the system. Such a measure indicates the total of all subsystem responses.

4-5. APPLICATION OF EQUALIZERS

Equalizers are used only when necessary to maintain or obtain the parameters established by the DCA for the various services performed by telephone networks.

a. Location. Equalizers are always placed at the receiving end or terminating point of a circuit. However, some lines cannot be equalized because of the facilities included in the wire circuit. For example, a long-distance trunk circuit containing a channel of telephone carrier cannot be equalized beyond the limits of the channel filter in the carrier terminal. On the other hand, a pair of wires should be capable of wide equalization, because the variation in reactance occurs relatively slowly with change in frequency, provided there is nothing else in the line. However, when loading coils or impedance-matching transformers are used, the equalizing limit is determined primarily by the frequency cutoff that results from combining these line devices.

b. Application to Telephone Carrier Terminal. The problems encountered in applying equalizers to telephone carrier terminals are illustrated in figure 4-6.

(1) The individual channels of a multichannel telephone carrier terminal are not normally equalized because equalizing is accomplished across the baseband of all combined channels at the receiving line terminations. As shown in A of figure 4-6, one equalizer serves all channels within the carrier terminal.

(2) An input loop circuit connecting to the channel of a telephone carrier terminal uses a wire pair which carries signals having an amplitude-frequency relationship. The line equalizer on the receive line cannot compensate for this condition. If equalization is important to the signals arriving at the loop input to the carrier channel, an equalizer will have to be installed at the termination of that loop.

(3) The equalizer will have to be placed at the terminating end of each pair of a four-wire circuit; it is not effective on a two-wire circuit because such a circuit passes signals in either direction. Therefore a hybrid coil must be used wherever equalization is required for a two-wire loop. This principle in illustrated in A of figure 4-6.

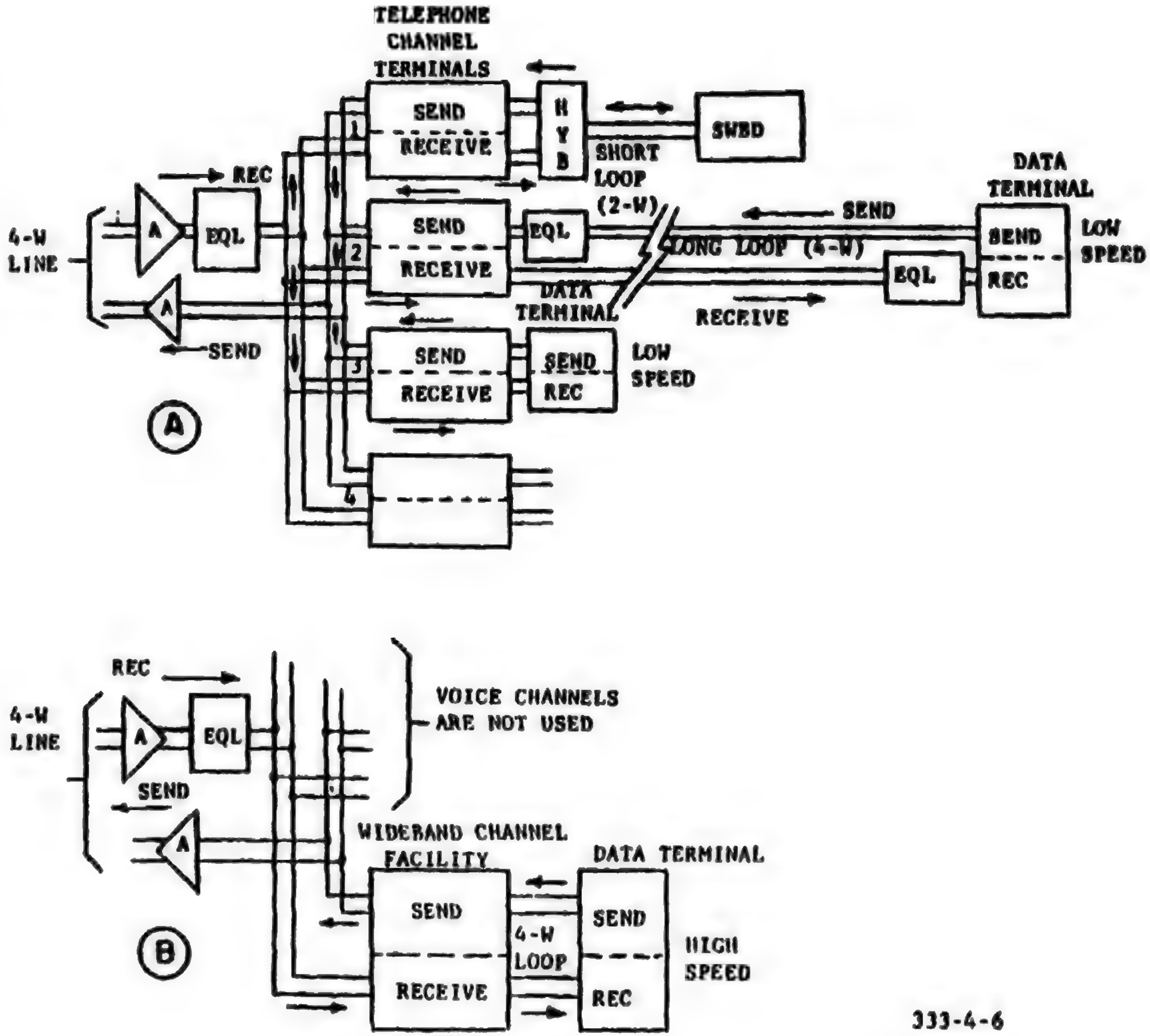


Figure 4-6. Application of equalizers to telephone carrier terminal.

c. Data Terminals. The data terminals in A of figure 4-6 are necessarily low-speed because their speed is limited primarily by the bandwidth of the filter in each telephone channel. High-speed data terminal operation can be secured by use of the full bandwidth capability occupied by four channels, as shown in B of figure 4-6. By doing this, we sacrifice the four independent channels to gain the benefit of one wideband facility. Under such an arrangement, a suggested method of operation is to place the data terminal near the telephone carrier terminal so that the same receive line equalizer serves both the telephone terminal and the data terminal for amplitude-frequency correction. With this system we avoid the necessity for modulation and use only amplifiers to pass the signal directly through the wideband channel facility. However, if envelope delay still exists, a combination equalizer will have to be used to compensate for the delay.

4-6. FUTURE DEVELOPMENTS

Conditioning telephone lines to pass data signals is relatively new. Equalizers now in use require considerable skill and training to operate, as well as much time for the adjustment procedure. Commercial services are developing highly sophisticated equalizing devices. One such device is semi-automatic in operation in that the operator first plots uncorrected circuit responses after which he sets controls in the exact opposite positions to selected points on the response curves. This procedure largely eliminates the time-consuming manual method now being used. Another new type of equalizer uses entirely automatic compensation under control of a computer. This combination makes possible self-analysis and correction of line characteristics. One disadvantage of the fully automatic system is that control is taken out of the hands of the operator. He is sometimes unable to take corrective action on the line because his attempt may cause the automatic features to readjust the equalizer away from the desired parameters.

STUDY EXERCISES

In each of the following exercises, select the ONE answer that BEST completes the statement or answers the question. Indicate your solution by circling the letter opposite the correct answer in the subcourse booklet.

1. The original purpose of an equalizer was to equalize a telephone circuit so as to minimize

   a. phase-frequency distortion.

   b. amplitude-phase distortion.

   c. harmonic-frequency distortion.

   d. amplitude-frequency distortion.

2. Assume that you observe a digital waveform on an oscilloscope. The fact that you find no change in waveshape of the received signal as compared with the transmitted signal indicates that the

a. envelope delay is maximum.

b. carrier frequencies are identical.

c. limiters are clipping the signal peaks.

d. harmonic-frequency relationship is maintained.

3. When the phase shift of a circuit is directly proportional to a change in frequency, the circuit exhibits the characteristic of

a. constant delay.

b. zero phase shift.

c. nonlinear rise of delay time.

d. linear relationship between amplitude and frequency.

4. Assume that a technician observes that a received digital pulse train is identical with the original waveform. He can deduce from this waveform that the

a. phase delay is proportional to frequency.

b. pulse delay time is proportional to frequency.

c. circuit has nonlinearity of delay with respect to frequency change.

d. higher order harmonics making up the digital pulses are all shifted the same amount.

5. In the simple amplitude-frequency equalizer shown in figure 4-2, midfrequency equalization is accomplished by adjustment of resistors R1 and R2. Midfrequency equalization results from the fact that these resistors will

a. correct for amplifier response.

b. broaden the response curves by varying Q.

c. change the resonant frequencies of the tuned circuits.

d. modify the loss characteristic of the telephone line proper.

6. Assume that you are equalizing a long telephone line. If you follow recommended procedure you will measure level at the

a. amplifier input.

b. equalizer input.

c. amplifier output.

d. equalizer output.

7. An ideal telephone pair has a "flat" amplitude-frequency response. This means that the

a. loss increases with a rise in frequency.

b. line has no loss anywhere along its length.

c. loss is constant over a specified band of frequencies.

d. amplitude variation is proportional to the change in frequency.

SITUATION

Assume that you have been directed by your station chief to determine the envelope-delay characteristics of circuits A and B, and to equalize them if necessary. Upon investigation, you find both circuits have identical envelope-delay characteristics like the uncorrected circuit curve in figure 4-7.

Exercises 8 through 11 are based on this situation.

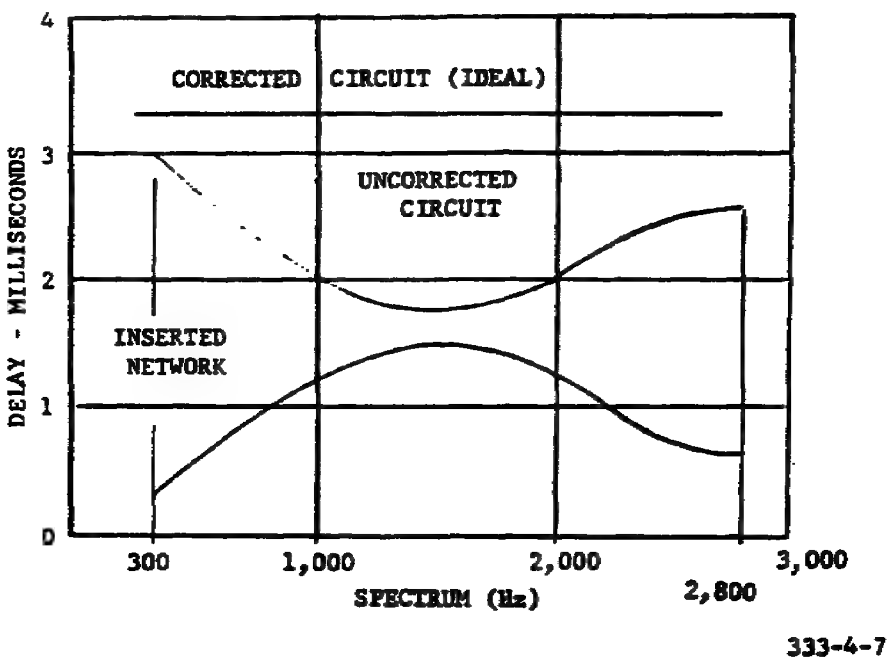


Figure 4-7. Delay equalized circuit characteristics.

8. Two equal delays of the uncorrected circuit occur at approximately the frequencies of

a. 300 and 1,000 Hz.

b. 300 and 3,000 Hz.

c. 1,000 and 2,000 Hz.

d. 2,000 and 3,000 Hz.

9. Upon completion of equalizing circuit A, the results are shown by the set of curves in figure 4-7. By analyzing these curves, the circuit conditioner can see that the

a. curve of the inserted network is identical in every way with that of the uncorrected circuit.

b. sum of delays of the inserted network and the uncorrected circuit varies with each test frequency.

c. delays in the corrected circuit all have the same value as shown by the straight-line characteristic.

d. difference between delays at each frequency always equals zero.

10. Circuit parameters for specified services are published by DCA. The parameters which specify limits of envelope delay relate

a. phase end delay.

b. delay and frequency.

c. amplitude and phase.

d. frequency and amplitude.

11. Assume that the parameters specified for circuit B include the following: 1.5 to 2.5 ms between 1,000 and 2,000 Hz. Will it be necessary to condition this circuit further?

a. Yes, because the uncorrected circuit and inserted network characteristics are not mirror images.

b. Yes, because the envelope delay of the uncorrected circuit is not linear with respect to frequencies above 2,000 Hz.

c. No, because no delays anywhere in the characteristic of the uncorrected circuit fall outside the 1.5- to 2.5-ms range.

d. No, because the characteristic of the uncorrected circuit shows that the circuit parameters are within requirements.

12. Assume that the input level at 1,000 Hz to the combination equalizer shown in figure 4-4 is -16 dbm, the output level needed is +5 dbm, and the total loss through the equalizer sections amounts to 10 db. How much of the 25-db amplifier gain must be utilized? (Disregard losses in the input and output transformers.)

a. 10 db

b. 16 db

c. 21 db

d. 25 db

13. One type of combination equalizer contains 12 equalizer sections. When a line has been properly equalized for amplitude-frequency distortion as well as envelope delay, the technician will find that

a. all equalizer sections are used.

b. line amplifiers are not needed because each section has its own amplifier.

c. the frequency response and delay characteristic curves are identical.

d. each of the equalizer sections used are tuned to a different frequency segment in the designated spectrum.

14. When a telephone circuit has to be conditioned for data transmission, everyone concerned must allow time for the process. One reason the procedure takes time is the

a. extent of coordination necessary between technicians.

b. large number of controls that must be preset.

c. voluminous records that must be maintained.

d. amount of experimenting needed.

15. A circuit conditioner must be aware that there is a limit to the ability of an equalizer to condition a long-distance telephone circuit. That limit is set primarily by the

a. amount of noise developed along the line.

b. total loss of all subsystems in combination.

c. cutoff frequency caused by the combining of subsystems.

d. harmonic distortion produced by overloaded repeating amplifiers.

CHECK YOUR ANSWERS WITH LESSON SOLUTION SHEET, PAGE 65, AND MAKE NECESSARY CORRECTIONS.

LESSON SOLUTIONS

SIGNAL SUBCOURSE 333........................Telephone System Characteristics

LESSON 1..................................Introduction to Telephone Systems

---

1. b--para 5, fig. 3
2. a--para 6b, c; 7
3. b--para 10c
4. a--Appendix B, para 6b
5. a--para 10b
6. e--para 12a

   74 x 20 = 1,480 ohms
7. b--para 14c
8. a--para 28a(6), (7); fig. 15
9. d--tables I, II; Appendix B, para 2a, b

When X_L is equal to X_C, the impedance is nearly equal to R. Tables I and II of Appendix A show that when F = 2,000 Hz, L = 0.006 henry and C = 0.000001 farad; X_L = 75.4 ohms and X_C = 79 ohms. Since the reactances are nearly equal, the circuit is largely resistive at approximately 2,000 Hz. The circuit impedance is therefore approximately equal to the value of R (157 ohm).

10. a--Appendix B, para 2
11. a--Appendix B, para 3, 5
12. a--para 8b; Appendix B, para 6b

---

13. c--Appendix B, para 8

14. d--Appendix B, para 9

15. d--Attached Memorandum, para 1-1

16. b--Attached Memorandum, para 1-1a(4)

17. c--Attached Memorandum, para 1-1b, fig. 1-1

18. a--Attached Memorandum, fig. 1-2

The 1,500-Hz line crosses the C-message weighting lose curve at approximately -2 db, F1A at -3 db, and 144 at -12 db. These loss values below 0 dbm give -3 dbm for F1A, -12 dbm for 144, and -2 dbm for C-message.

19. c--Attached Memorandum, para 1-2b(2)

20. d--Attached Memorandum, para 1-3c

EXTENSION COURSE OF THE US ARMY SIGNAL SCHOOL

LESSON SOLUTIONS

SIGNAL SUBCOURSE............................ Telephone System Characteristics

LESSON 2.................................... Network Losses and Gains.

---

1. b--para 2-1d(5), example 1

   Attenuator loss = 35 db - 27 db = 8 db.

2. b--para 2-2a, fig. 2-4

   The power ratio represented by 3 db is 1/2 or 2/1. Since level drops 3 db to cutoff frequency, 1/2 of 8 mw is 4 mw.

3. a--para 2-2b

4. c--para 2-2b, table II

5. a--para 2-2b, table II

6. a--para 2-2d, example 2; fig. 2-1

   The power level of +3 dbm is 2 mw. Since each channel signal is 1/4 of that total power, the power level in each signal is 2/4, or 1/2 mw.

7. d--para 2-1c

8. c--para 2-1f

9. c--para 2-1d, 2-3c

   The 20-db loss in the attenuator drops the signal level to -32 dbm. The 10-db loss in the equalizer drops the signal level to -42 dbm. The 42-db fixed-gain amplifier raises the level to 0 dbm. The 6-db pad reduces the signal level to -6 dbm. In summation, the total loss is 36 db and the amplifier gain is 2 db, making the overall circuit gain 6 db. With the input signal of -12 dbm and circuit gain of 6 db, the final output level is -6 dbm.

---

10. c--para 2-3f

The TLP level is -3 dbm. This level is therefore 0 dbm0. The 6-db pad determines that the signal entering the equipment will be at a relative level of 6 db below 0 dbm0, or -6 dbm0.

11. d--para 2-4

12. a--fig. 2-1

13. b-table I

A gain of 6 db is equivalent to a power ratio of approximately 4 times (3.981).

14. c--para 2-5c

Noise level of -50 db is 40 db above -90 db (0 dbrn), so it can be given as +40 dbrn.

15. c--para 2-5, example 2

Noise level of -50 dbm is 53 db below the TLP of +3 dbm. Noise level with respect to the TLP is therefore -53 dbm0.

16. d--para 2-6c

17. b--para 2-6c

The nominal loss for an LD trunk is 3 db. The receive level is therefore 3 db less than the send level of -3 dbm, or a level of -6 dbm.

18. d-para 2-1c(4)

19. a--para 2-6d(1)

-3 dbm + 6 db = +3 dbm.

20. b--para 2-6d(2), fig. 2-8

EXTENSION COURSE OF THE US ARMY SIGNAL SCHOOL

LESSON SOLUTIONS

SIGNAL SUBCOURSE 333........................Telephone System Characteristics

LESSON 3...................................Echo Suppressors and Compandors

---

1. c--para 3-1a

2. d--para 3-1b

3. c--para 3-2

4. b--para 3-2b(1)

5. a--para 3-2b(1)

6. d--para 3-2b(3)

A hybrid coil circuit always introduces at least 4-db loss. A signal of -2 dbm entering the 2-W side of the hybrid coil circuit will leave the send line at a level approximately 4 db below -2 dbm, or -6 dbm.

7. b--para 3-2c

8. a--para 3-3a(3)

9. b--para 3-3b(1), (2)

10. c--para 3-3c

11. d--Appendix C, page C-2

12. c--Appendix C, page C-11

13. a--Appendix C, page C-12

14. c--Appendix C, page C-13

15. d--Appendix C, page C-13

16. c--Appendix C, page C-13

17. a--Appendix C, page C-15

18. d--Appendix C, page C-16

With a compression ratio of 2:1, the intensity range between the compressor and expandor is one-half the companding range.

1/2 x 50 db = 25 db.

19. a--Appendix C, page C-16

20. d--Appendix C, page C-16

EXTENSION COURSE OF THE US ARMY SIGNAL SCHOOL

LESSON SOLUTIONS

SIGNAL SUBCOURSE 333...................... Telephone System Characteristics

LESSON 4................................ Equalization

---

1. d--para 4-1a(3)

2. d--para 4-1b(2)

3. a--para 4-1b(3)

4. a--para 4-1b(3), (4)

5. b--para 4-2b(1), (2); fig. 4-2

6. c--para 4-2a(3)

7. c--para 4-2a

8. c--para 4-3a, fig. 4-7

9. c--para 4-3a, fig. 4-7

10. b--para 4-3a, fig. 4-7

11. d--para 4-3a(3)

Parameter limits lie between 1.5 and 2.5 ms. The maximum variation of the uncorrected circuit curve lies between 1.8 and 2.0 ms over the range of 1,000 to 2,000 Hz, well within limits. The circuit conditioner need take no action to equalize the circuit.

12. c--para 4-3c, fig. 4-4

The 10-db fixed-gain amplifier will compensate for the 10-db loss in the equalizer sections. The 25-db amplifier must raise the input level of -16 dbm to the output level of +5 dbm, a total of 21 db. Therefore, 21-db gain of the 25-db amplifier is needed.

13. d--para 4-3b, c

14. d--para 4-4

15. c--para 4-5a

APPENDIX A

THE TELEPHONE
TRANSMISSION SYSTEM
SSTS 53002A

OBJECTIVES

1. To explain the makeup of transmission systems.

2. To show you how a telephone transmission system works.

3. To explain the meaning of voice power.

4. To tell you how resistance, leakage, inductance, and capacitance affect the transmission of voice power.

This information sheet supersedes SSTS 53002, The Telephone Transmission System.

TABLE OF CONTENTS

1. INTRODUCTORY INFORMATION

a. The purpose of a telephone system is to transmit voice frequencies ranging from 200 to 4,000 Hertz (Hz). Doing this becomes more and more difficult as we increase the length of the transmission line. The longer we make the line, the less sound we receive at the far end. The line actually limits the distance over which we can transmit.

b. To understand why this happens you have to know more about a telephone transmission system. That's the purpose of this information sheet: to tell you what a transmission system is and to show you how it works.

2. THE PURPOSE OF TRANSMISSION SYSTEMS

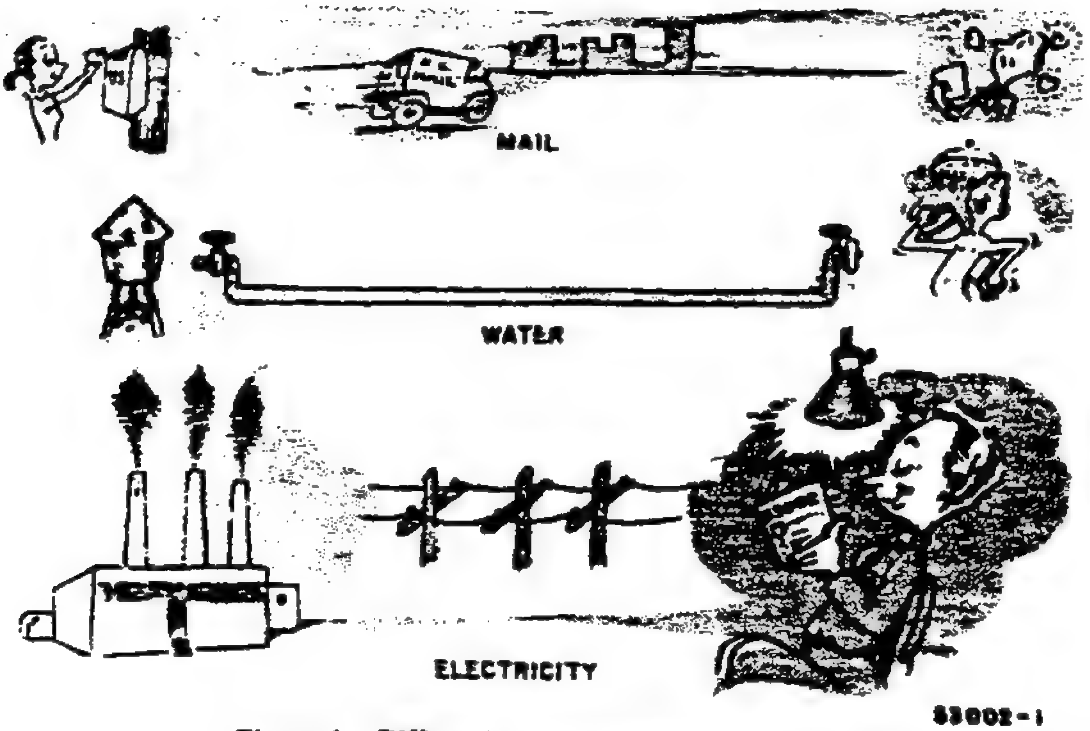
a. Every day you come across many different transmission systems. Here are some examples that you're familiar with (fig. 1): electric, water, mail delivery, bus transportation, trucking and the telephone.

b. All of these systems have one thing in common. They transmit something from one place to another. Of course, each one transmits something different. Still, each system must have the same three main parts to be able to do its job.

3. PARTS OF A TELEPHONE TRANSMISSION SYSTEM

a. Any transmission system has to have a source of energy, a transmission medium, and a receiving device. These are rather high sounding terms for some familiar things. Figure 2 on page 4 gives you an idea of what these things are and why they are needed.

b. The source of energy is needed because it provides what we transmit. The transmission medium is needed to carry what we transmit. Finally, the receiving device is needed to receive what we transmit and change it into some useful form. In the first example shown, the "receivers" convert what they receive into action. In the second, the telephone receiver changes what it receives into sound. Now, let's look at the parts of a telephone system and see how they work.



53002-1

Figure 1. Different transmission systems.

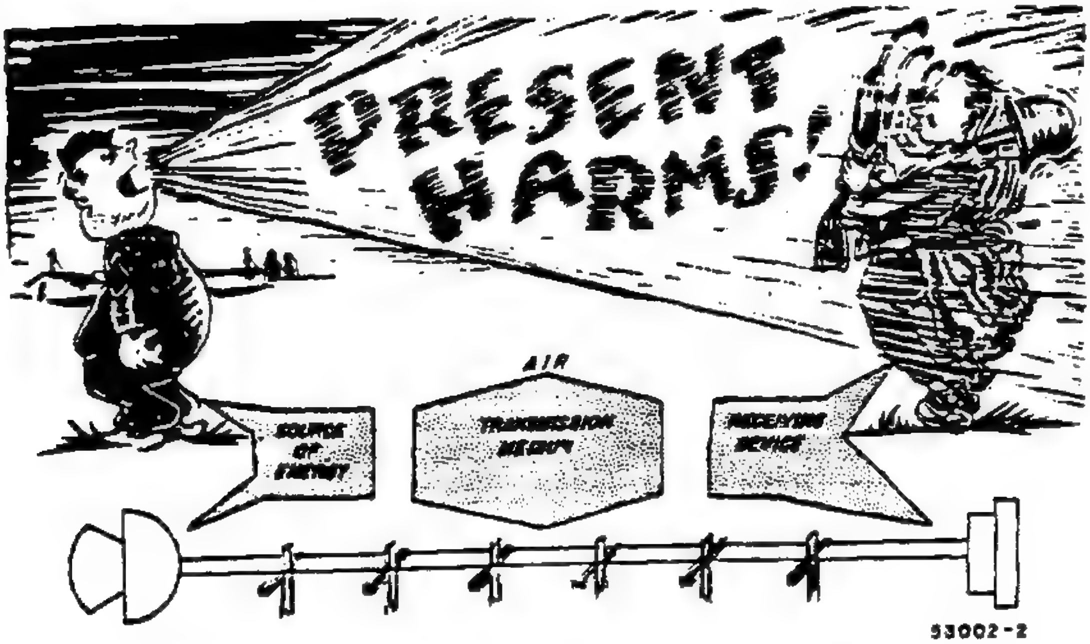


Figure 2. A transmission system needs three parts.

4. SOURCE OF ENERGY IN A TELEPHONE SYSTEM

In a telephone system, the source of energy is the transmitter circuit. It consists of a transmitter, a battery, and an induction coil. This circuit changes sound energy into electrical energy and transfers the electrical energy to the line. In an earlier information sheet you learned how this circuit works. Here we briefly review the circuit operation.

5. SOUND ENERGY IS CONVERTED TO ELECTRICAL ENERGY

a. The transmitter circuit is connected as shown in figure 3. The transmitter acts as an adjustable resistor and controls the current flow from the battery. When no sound is striking the transmitter, steady direct current (dc) flows in the circuit.

b. When sound waves strike the transmitter, the diaphragm bends in and out at the frequency of the sound wave. This in-out movement of the diaphragm makes the resistance of the carbon granules decrease and increase at the same frequency as the sound wave. A flow of fluctuating dc results. And, since the fluctuating dc flows through the primary of the induction coil, alternating current (ac) voltage is induced across the secondary.

c. The secondary of the coil is connected to the line. And the induced ac voltage is, therefore, applied directly across the line. The induced ac voltage causes a flow of alternating current in the line and through the receiver at the other end. This alternating current is the same frequency as the original sound wave that started the action.

d. The ac voltage across the line and the current flowing in the line make up the electrical energy that we call voice power.

6. STANDARD AMOUNT OF TRANSMITTED VOICE POWER

a. When a person talks into a telephone in a clear, loud voice, a very small amount of electrical energy is produced. This energy is in the form of ac voltage and current. Both of these (voltage and current) are so small that we usually don't speak of them by themselves. Instead, we speak of the combination of the voltage and current, which is power.

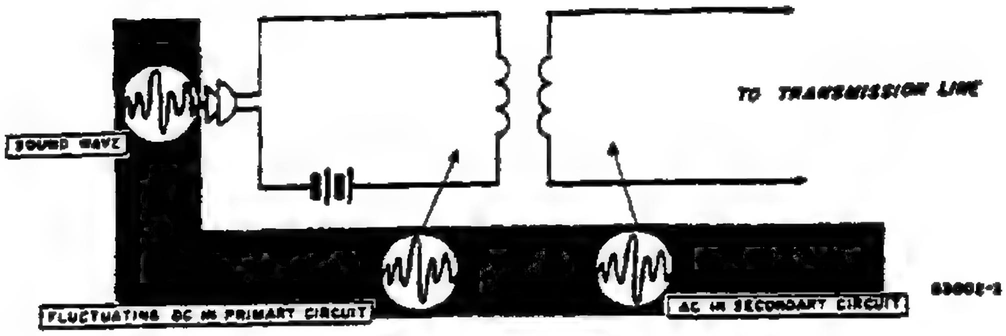


Figure 3. Telephone transmitter circuit.

b. You remember from your study of Ohm's Law that power is the product of voltage and current. In other words, power in watts can be found by multiplying voltage by current (P = E x I). Working this formula out for the power produced by a voice sound in a telephone gives us something like this:

(1) The voltage across the induction coil secondary is about 1 volt.

(2) The current caused by this voltage is about .001 ampere (1 milliampere).

(3) And since power in watts is:

P = Voltage (E) x Current (I)

P = 1 volt x .001 ampere

P = .001 watt (1 milliwatt)

c. The voltage and current values given in the formula are approximate. These values vary according to the loudness of the voice. But the value of 1 milliwatt is considered as the standard for voice power produced in a telephone. And you'll be dealing with this standard milliwatt from now on.

7. VOICE POWER IS CONVERTED INTO ELECTRICAL POWER

As we said before, your job is to transmit sound. But you hardly ever deal with sound in its natural form. When you get the sound, it's usually in the converted form of electrical power. And this power -- this 1 milliwatt -- is extremely small. It's so small that it would take 60,000 milliwatts to light one 60-watt lamp bulb. Yet, despite its smallness, this milliwatt of voice power has to go from one end of a transmission line to the other. And you have to see that it gets there.

8. TRANSMISSION MEDIUM AFFECTS VOICE POWER

a. The transmission medium of a telephone system is the part that carries voice power from the transmitter to the receiver. The transmission medium is the telephone line.

b. A telephone line (fig. 4 on page 6) consists of two wires separated by some type of insulation. This is true whether the wires are in a cable or out in the open. Bare open wires, like those you see strung on telephone poles, are insulated from each other by air, while wires in a cable are insulated with a paper, rubber, cotton or other type of covering.

c. Regardless of type, however, all lines perform the same job. They all affect the voice power in about the same way. By this we mean that all telephone lines reduce the amount of voice power that we are trying to transmit. In other words, you can never expect to receive the same amount of voice power at the receiver as you start out with at the transmitter. The telephone line always reduces the voice power that it is carrying. This reduction of voice power is called attenuation.

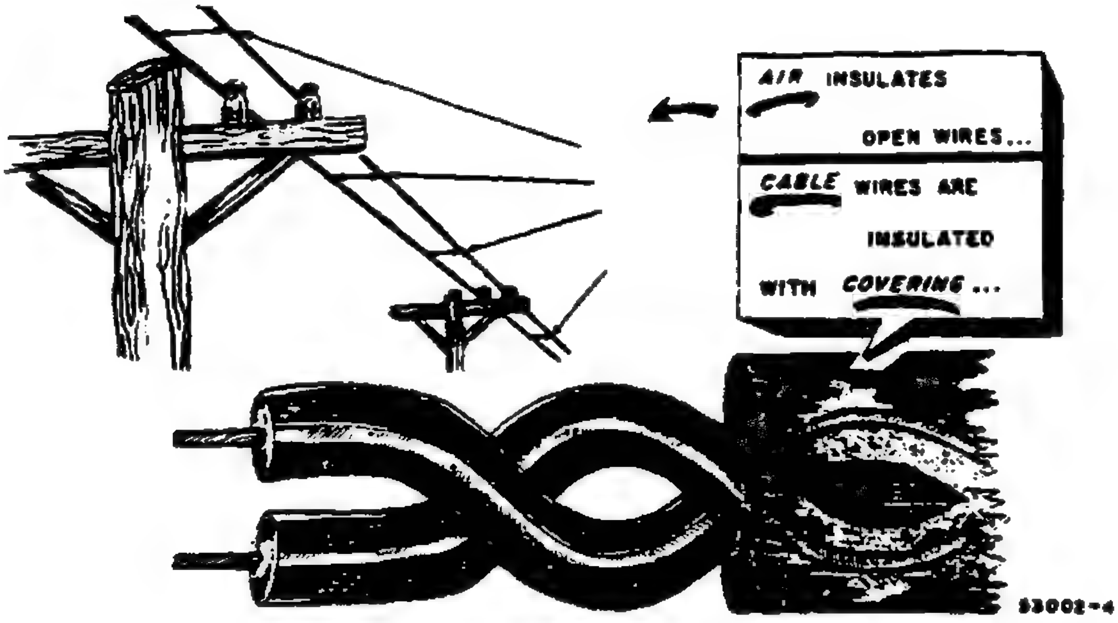


Figure 4. Types of telephone wires.

9. DEFINITION OF ATTENUATION

a. Attenuation comes from the verb attenuate which means to reduce, cut down, lessen, thin out, decrease, or weaken. Although we don't hear the word used in everyday language, it could be used in talking about everyday things. For instance, it would be correct to say, "If you do not read this text carefully, your grade for this lesson may be greatly attenuated."

b. In your work, you'll see and hear these words used very often:

(1) ATTENUATE -- which means REDUCE or CUT DOWN

(2) ATTENUATION -- which means REDUCTION

(3) ATTENUATOR -- which is a type of circuit that REDUCES or CUTS DOWN.

c. Know what these words mean when you come across them. Begin to use them yourself. They are as much a part of your work as the words volts and ohms.

10. TELEPHONE LINES ATTENUATE VOICE POWER

a. Making this statement is something like asking, "Why does a road cause automobile tires to wear out?" Everyone knows that it's the constant friction of the tires against the road surface that wears them out. It's just about the same with the telephone line. The constant friction of the wires wears away the voice power.

b. There are two different kinds of friction in these examples. On the road it's mechanical friction. In the telephone line, it's electrical friction caused by Resistance (R), Leakage (G), Inductance (L), and Capacitance (C) (fig. 5).

c. These four are present in all telephone lines. Because of this, they are called the four properties or the four characteristics of a transmission line. Each one has a different effect on voice power. We'll consider each one separately in the paragraphs that follow.

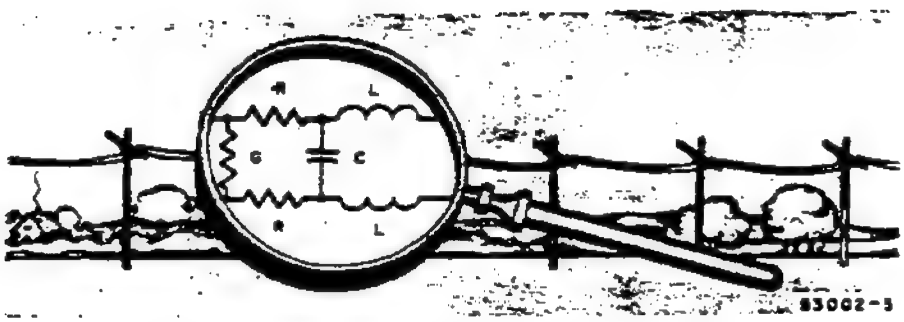


Figure 5. Looking at a telephone line more closely.

11. RESISTANCE (R) OF CONDUCTORS

a. All conductors have resistance. A piece of wire no larger than a pencil lead has a definite amount of ohmic resistance. Even though this piece of wire is extremely small, you can still measure the resistance.

b. Here's an example. One type of wire used in telephone lines has a resistance of about .00066 ohms per inch (fig. 6). This is a very small value and you might think we could forget about it. But suppose you put about 63,330 inches of this wire together into one length. It would equal one mile of wire. The resistance for this mile would be: .00066 ohms times 63,330 inches, which is about 43 ohms.

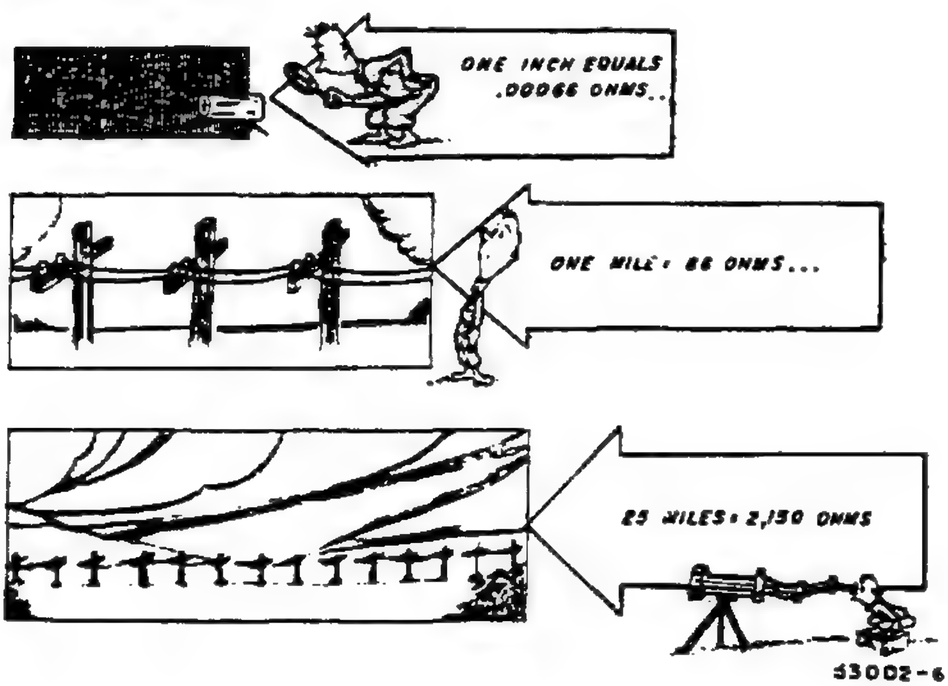


Figure 6. Telephone lines have resistance.

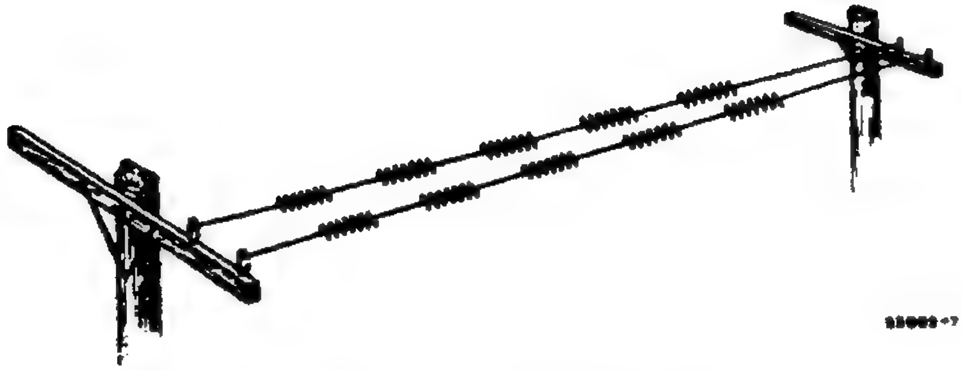


Figure 7. Line wires act as a series-resistance circuit.

12. USING WIRE WITH RESISTANCE IN A TELEPHONE LINE

a. Now suppose you wanted to use the same type of wire to construct a one-mile telephone line. You'd have to get two pieces one mile long, because a telephone line needs two wires. The resistance of each wire would be 43 ohms. This means that the total resistance of the one-mile line would be twice 43 ohms or 86 ohms. Remember, this 86 ohms is the resistance for only one mile of line. Lines, however, are much longer than this. If this same type of wire is used in a 25-mile circuit (which is more like it) the resistance would be 2, 150 ohms (86 ohms x 25 miles).

b. Simply stated, the two wires of a telephone line (fig. 7) act as two very long resistors whose resistance increases as the length of the line increases. The transmitter circuit is connected to one end of these "resistors," and the receiver circuit is connected to the other end. These "resistors" attenuate the voice power. But before we explain exactly how this attenuation takes place, we want to talk about another type of resistance that is present in a telephone line. This is the resistance between the two wires which causes leakage (G).

13. LEAKAGE (G) THROUGH INSULATORS

a. There are no perfect insulators. All materials, including rubber, paper, wood, glass, and air are really conductors. Of course, they are very poor conductors; that's why they are called insulators. And, for many practical purposes, these materials act as perfect insulators -- but not in a telephone line.

b. In a line, insulating materials are used to prevent current from flowing between the wires (from one wire to the other). But because these materials are not perfect, they can't stop all the current flow. And, therefore, the insulators themselves provide a high resistance path for current flow between the wires. This high resistance is called insulation resistance. And the current flowing through the insulation is called leakage. The symbol for leakage is the capital letter G. Leakage occurs in all types of telephone lines, whether they are bare open wires or insulated cable wires.

14. LEAKAGE IN OPEN WIRE LINES

a. The insulation used between the wires of open wire lines is air -- a fairly good insulator. Nevertheless, it provides a path for current flow. The air between the wires acts as a lot of resistors connected in parallel between the wires along the full length of the line (fig. 8).

b. In the figure, you also see resistors connected from the two wires to ground. These resistors represent the path through which small amounts of current leak to ground. Leakage to ground happens in different ways:

(1) Current leaking directly through the air down to the earth.

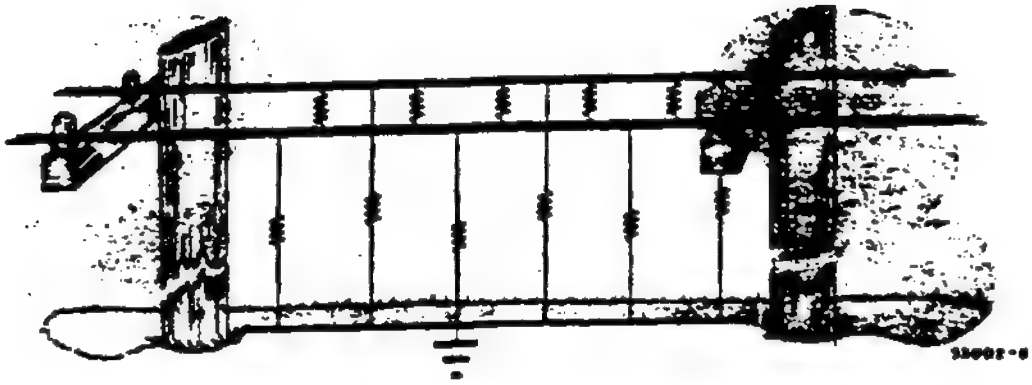


Figure 8. Air acts as parallel resistors.

(2) Leakage at points where the wires are attached to cross-arms of telephone poles. Current leaks from the wire, down through the pole and into the ground.

(3) Leakage of current through tree branches (which may touch or be near the wires) down through the tree trunk to ground.

c. An important point to remember about leakage in open wire lines is that weather conditions affect it greatly. In damp or wet weather, the air between the wires becomes a much poorer insulator. The insulation resistance goes down and there is more leakage. This doesn't happen in cables because the wires are sealed in where moisture can't affect them. But we still get leakage in cables.

15. LEAKAGE IN CABLES

a. In cables, the wires are close together. The insulation around each wire touches the insulation of another wire. This insulation has resistance (fig. 9 on page 10) just like the air in open wire lines. And this resistance is also very high -- usually higher than the resistance of air. Therefore, we still get leakage between the wires. And we still get some leakage to ground.

b. Leakage to ground happens in cables because there is usually a metal covering around the cable, which is connected to ground. In some cables, this covering is on the outside and it's made of lead. In others, the metal covering is in the form of a metal braid (a basket weave) which is under an outside rubber or plastic covering. This last one, by the way, is the type of cable that you'll use with carrier equipment (spiral-four cable).

16. PRACTICAL IDEAS ABOUT LEAKAGE

a. In discussing leakage, we've said that the insulation resistance which causes leakage is very high. This phrase "very high" doesn't really say much. So to give you an idea of how high insulation resistance is, here are a few facts: Insulation resistance is so high that it is measured in megohms (millions of ohms), and it decreases as the length of the line increases.

b. For example, the insulation resistance of the spiral-four cable we mentioned earlier is 200 megohms (200 million ohms) for one mile. But the insulation resistance goes way down to 8 megohms when we increase the length of the cable to 25 miles. Each mile is like a 200-megohm resistor connected in parallel. By using Ohm's Law we find that twenty-five 200-megohm resistors connected in parallel is equal to 8 megohms.

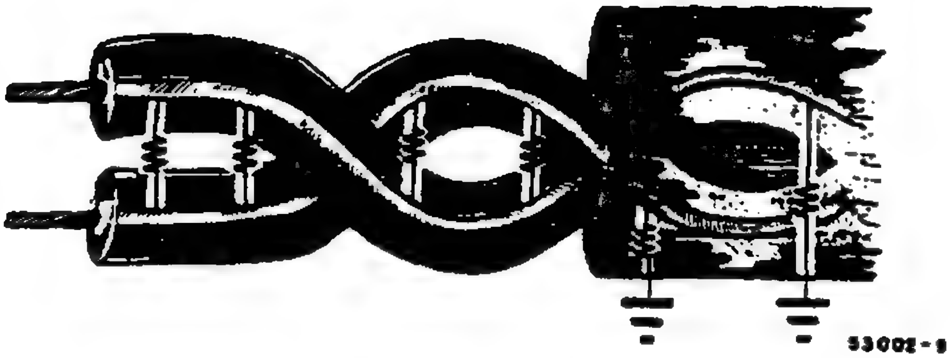


Figure 9. Leakage in cables.

c. You can see that leakage will increase as the line is made longer. It's as simple as this: Longer line = lower insulation resistance = more leakage.

d. Now here's the important thing about leakage. The current leaking between the wires and to ground is part of voice power. It's part of the 1 milliwatt that you're trying to send to the receiver. And this loss caused by leakage, plus the loss caused by resistance of the wires, is attenuation.

17. RESISTANCE AND LEAKAGE CAUSE ATTENUATION

a. Now you know what is meant by the resistance and leakage properties of a line. The next question is: How do these properties attenuate the voice power?

b. Assume that you have sent out from a transmitter 1 mw (1 milliwatt) of voice power. Assume further that this 1 mw is made up of 1 volt and 1 milliampere (ma.) (1 volt x .001 ampere = .001 watt). At the far end of the line you would like to receive the same amount of voltage, current and power as shown in figure 10. You know, however, that this is impossible because of the resistance of the wires and insulation resistance. Instead of getting the same amount of power out, you'd get something like what you see in figure 11.

18. RESISTANCE CAUSES A VOLTAGE DROP

The resistance of the wires causes the voltage applied at the transmitting end to drop. Every tiny millionth of an inch of wire along the way causes the voltage to drop a little more. This means that the voltage across the receiver is less than the voltage applied at the transmitter.

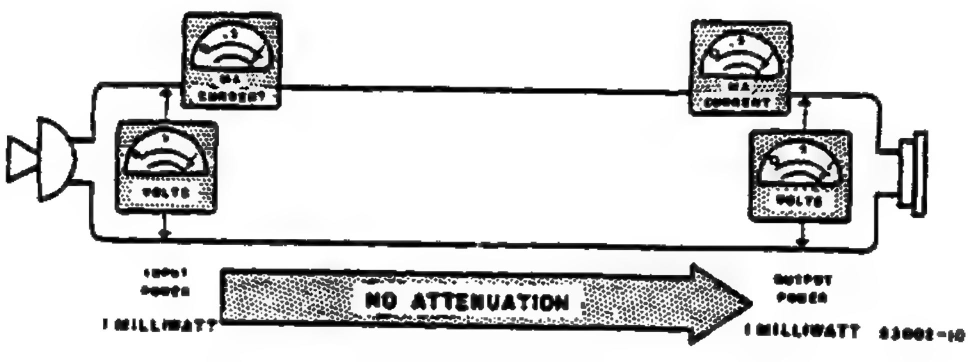


Figure 10. Ideal transmission system.

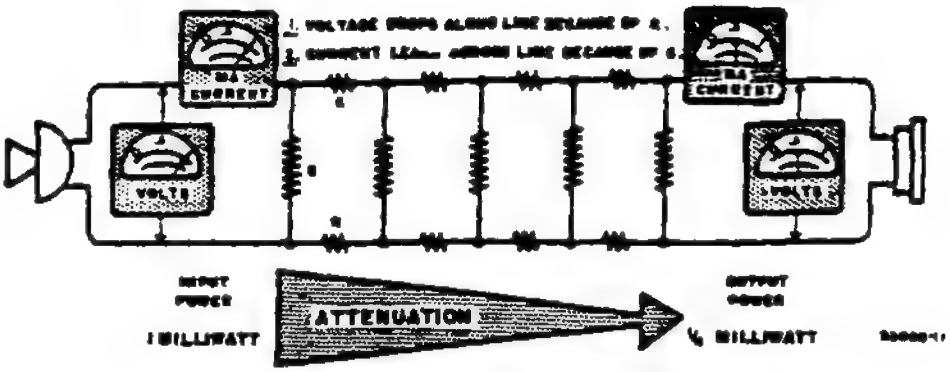


Figure 11. Power received is less than power sent.

19. LEAKAGE RESULTS IN A CURRENT LOSS

Figure 11 also shows that all of the 1 ma. of current supplied from the transmitter does not reach the receiver either. All along the line, current leaks through the insulation and returns to the transmitter. This leakage-current does not reach the receiver. So, flowing through the receiver, there is less current than the 1 ma. that started out from the transmitter.

20. LESS VOLTAGE AND LESS CURRENT MEANS LESS POWER.

a. The current and voltage at the receiving end are less than the transmitted voltage and current. This results in less voice power at the receiver, since $P = E \times I$. You can see this in figure 11 where 1 mw is sent but only 1/4 mw is received. Three-fourths or 75 percent of the voice power is lost in this circuit because of resistance and leakage. That's a lot of attenuation. Still, it's only part of the total attenuation that we get in a line.

b. The other two line properties, inductance and capacitance, also attenuate the voice power.

21. INDUCTANCE (L) AND CAPACITANCE (C) CAUSE ATTENUATION

a. Because of these two properties, a line causes attenuation which is not the same for all frequencies. The attenuation gets greater as the transmitted frequencies go higher. This means that a line reduces the voice power of people who speak at high frequencies more than those who speak at low frequencies. Also, when people change from low to high frequencies while telephoning, they are not heard as well at the high frequencies, even though they talk just as loud both times. This is shown graphically in figure 12 on page 12.

b. The graph on the left shows that all frequencies from 200 to 4, 000 Hertz are transmitted at the same power level of 1 mw. The graph on the right shows that all frequencies are received at different power levels. You'll notice that less power is received for each higher frequency. This is called unequal attenuation, and results in distortion. In other words, the line does not give faithful reproduction because of inductance and capacitance. Why do these two properties cause unequal attenuation? Let's study them one at a time and find out.

22. INDUCTANCE (L) OF TELEPHONE LINES.

a. A line has inductance because of the ac voice currents flowing through the wires. Here's what happens:

(1) Ac voltage from the transmitter causes ac voice current flow in the line

(2) The ac voice current flow produces a changing magnetic field around the wires.

(3) The changing magnetic field induces voltages into the wires all along the line.

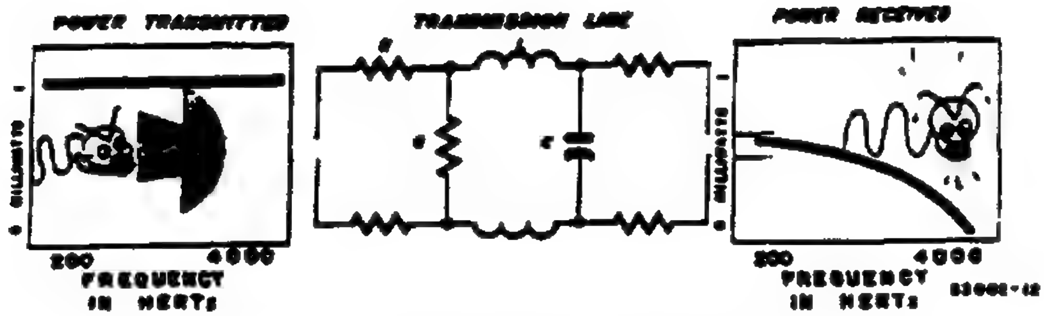


Figure 12. Inductance and capacitance cause unequal attenuation.

(4) The induced voltages oppose the applied voltage that is causing the current flow.

(5) This results in gradual reduction of voltage along the line, as shown in figure 13.

(6) Less voltage gives less current all along the line too, since current is a direct result of voltage.

b. What this all adds up to is a loss of voice power. Less voltage and less current reach the receiver and, therefore, less power is delivered to the receiver.

23. VOLTAGE, CURRENT, AND POWER LOSSES INCREASE WITH FREQUENCY

This is easy to see if you remember this: The opposing voltages induced into the wires (as a result of voice current flow) become greater as the magnetic field speeds up its movement. And since this happens when voice current flows at a higher frequency, there is naturally more opposition to the applied voltage at this time. So, for each higher frequency transmitted, there is less voltage, less current, and less power all along the line.

24. INDUCTIVE REACTANCE IS DETERMINED BY A FORMULA

a. The opposition caused by inductance is called inductive reactance (X_L). This is expressed in ohms just like dc resistance. To find out how much inductive reactance there is in a circuit, we use the formula $X_L = 2\pi FL$. You've seen and used this formula before. And you know that as F (frequency) is increased, X_L also increases as long as L (inductance) is not changed.

b. The formula has been worked out for nine different frequencies in Table I. There it's plain to see that with an inductance of .006 henries, the opposition (X_L) increases from 7.5 ohms at 200 Hertz all the way up to 150.7 ohms at 4,000 Hertz. Now, since you know that greater opposition means greater loss, you can see why inductance causes unequal attenuation.

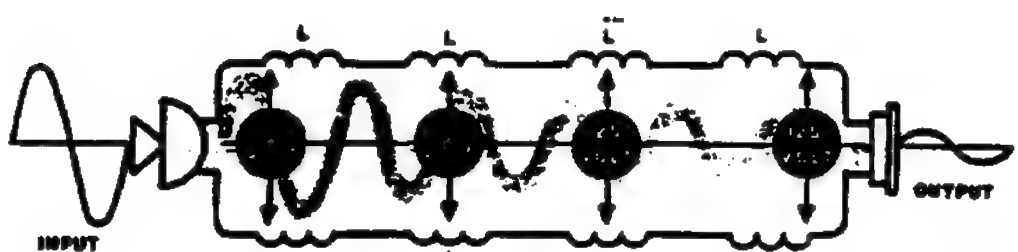


Figure 13. Inductance reduces voltage.

TABLE I

X_L (Inductive Reactance)		2π	x	F (Frequency)	x	L (Inductance)
7.5 ohms	=	6.28	x	200 Hertz	x	.006 henries
11.3 ohms	=	6.28	x	300 Hertz	x	.006 henries
15.1 ohms	=	6.28	x	400 Hertz	x	.006 henries
37.6 ohms	=	6.28	x	1,000 Hertz	x	.006 henries
75.4 ohms	=	6.28	x	2,000 Hertz	x	.006 henries
113.0 ohms	=	6.28	x	3,000 Hertz	x	.006 henries
120.6 ohms	=	6.28	x	3,200 Hertz	x	.006 henries
131.8 ohms	=	6.28	x	3,500 Hertz	x	.006 henries
150.7 ohms	=	6.28	x	4,000 Hertz	x	.006 henries

25. TELEPHONE LINES HAVE CAPACITANCE (C)

a. A capacitor consists of two conductors separated by some type of insulation. This is what you have in a telephone line, two long conductors separated by insulation. This holds true for open wire as well as cable lines.

b. The capacitance in a line (fig. 14), as you can see, is present between the wires for the full length. It's just as if there were small capacitors connected between the wires at every point along the line. This is what we mean by the capacitance (C) property of a line.

c. You already know that capacitance (C) causes unequal attenuation of voice power. Now we want to find out how this happens.

26. HOW CAPACITANCE (C) AFFECTS VOICE POWER

a. Capacitance provides a path for current to flow between the wires of the line. This path offers less opposition as the frequency increases.

b. In other words, as the frequency of the voice currents being transmitted increases, more and more current is shunted from one wire to the other. Of course this doesn't mean that current flows through the capacitance path. It means that since voice currents are ac, they effectively flow through the capacitance path just as ac effectively flows through any capacitor.

c. Now since this shunting effect of capacitance becomes greater with a higher frequency, less current is delivered to the receiver at the higher frequency. We can say, then, that capacitance causes attenuation which increases as the frequency rises.

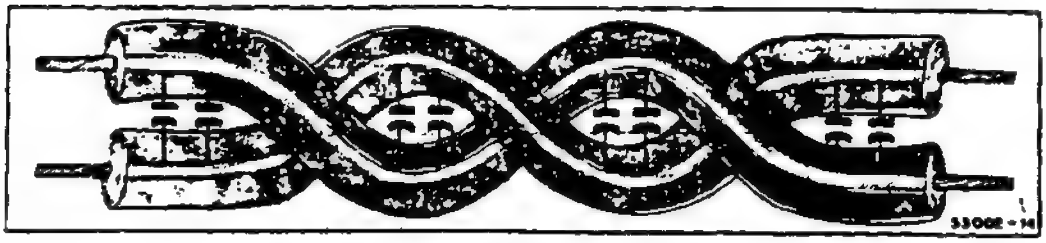


Figure 14. A telephone line has capacitance.

TABLE II

X_C (Capacitive Reactance)	=	$\dfrac{1}{2\pi \times F \text{ (Frequency)} \times C \text{ (Capacitance)}}$
796 ohms	=	$\dfrac{1}{6.28 \times 200 \text{ Hertz} \times .000001 \text{ farad}}$
531 ohms	=	$\dfrac{1}{6.28 \times 300 \text{ Hertz} \times .000001 \text{ farad}}$
398 ohms	=	$\dfrac{1}{6.28 \times 400 \text{ Hertz} \times .000001 \text{ farad}}$
159 ohms	=	$\dfrac{1}{6.28 \times 1,000 \text{ Hertz} \times .000001 \text{ farad}}$
79 ohms	=	$\dfrac{1}{6.28 \times 2,000 \text{ Hertz} \times .000001 \text{ farad}}$
53 ohms	=	$\dfrac{1}{6.28 \times 3,000 \text{ Hertz} \times .000001 \text{ farad}}$
50 ohms	=	$\dfrac{1}{6.28 \times 3,200 \text{ Hertz} \times .000001 \text{ farad}}$
46 ohms	=	$\dfrac{1}{6.28 \times 3,500 \text{ Hertz} \times .000001 \text{ farad}}$
40 ohms	=	$\dfrac{1}{6.28 \times 4,000 \text{ Hertz} \times .000001 \text{ farad}}$

27. CAPACITIVE REACTANCE IS DETERMINED BY A FORMULA.

a. The capacitance path for current flow between the wires of a line decreases its opposition as the frequency rises. This opposition is called capacitive reactance (X_C). And the amount of opposition (X_C) to currents of any frequency can be found by the formula:

$$X_C = \frac{1}{2\pi FC}.$$

b. You have seen and used this formula. And you know, that if C (capacitance) is held constant while F (frequency) is increased, X_C will decrease. Examples of this are given in Table II. In the table the capacitance is held at 1 mf (.000001 farad) while the frequency is varied from 200 to 4,000 Hertz.

c. From the table you can see that capacitive reactance (X_C) reduces its opposition from 796 ohms at 200 Hertz to only 40 ohms at 4,000 Hertz. This means, of course, that more current is effectively shunted across the line at 4,000 Hertz than at 200 Hertz, and, therefore, less current reaches the receiver at 4,000 Hertz. Less current naturally means less power. You can see, therefore, why capacitance causes attenuation which increases as the frequency rises.

28. THE FOUR ELECTRICAL PROPERTIES OF TELEPHONE LINES

a. You realize now that the transmission medium for a telephone system is not as "innocent" as it looks. The simple wires that do the job have more in them than meets the eye. And these hidden properties cause attenuation and what's even worse, unequal attenuation. Let's briefly review these important line properties again.

(1) First of all, remember that you're transmitting 1 milliwatt of voice power. This is applied at the transmitter in the form of small amounts of voltage and current.

(2) The telephone line which carries this 1 milliwatt is really a long circuit (fig. 15) composed of the following:

(a) Series RESISTANCE (R).

(b) Parallel (insulation) resistance which causes LEAKAGE (G).

(c) Series INDUCTANCE (L), and

(d) CAPACITANCE (C) between the wires.

(3) Each one of these properties reduces the voice power.

(4) RESISTANCE causes the applied voltage to drop all along the line, resulting in less voltage, and less power at the receiver.

(5) LEAKAGE is the current that flows from one wire (through the insulation) to the other wire. This happens all along the line too. The current that leaks is not delivered to the receiver. There is, therefore, less current and less power at the receiver.

(6) INDUCTANCE causes opposition (X_L) that reduces the applied voltage all along the line. This opposition increases as the frequency rises ($X_L = 2\pi FL$). This gives increasingly less voltage, less current and less power at the receiver as the frequency rises.

(7) CAPACITANCE effectively shunts current from one wire to the other all along the line. More current is shunted at higher frequencies because the opposition to current flow (X_C) decreases as the frequency rises ($X_C = \frac{1}{2\pi FC}$). This causes increasingly less current and less power at the receiver as the frequency rises.

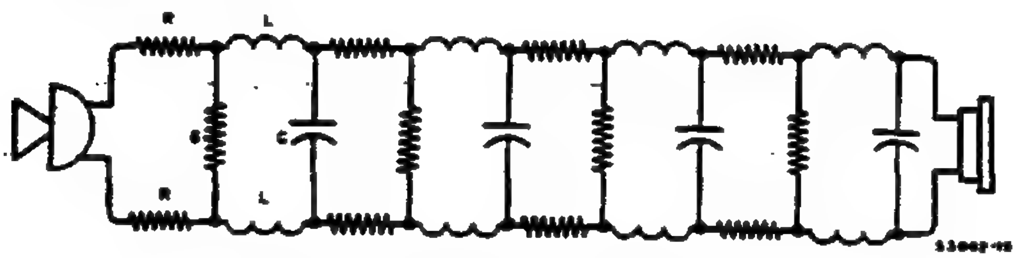


Figure 15. Telephone line properties.

b. Finally, then, we say that the line properties (R, G, L, C) attenuate the voice power. And the attenuation increases as the frequency rises. In other words, the line reduces the loudness of all sounds transmitted and this reduction increases as sounds get higher in frequency.

29. ONE MILLIWATT OF POWER IS NEEDED TO TRANSMIT SOUND

All that's been said about attenuation probably has you wondering how you ever hear anything over the telephone. Well, the situation isn't as bad as it may seem. Even though we start out with only 1 milliwatt, we can lose quite a bit and still get sound from the receiver.

30. MAXIMUM ALLOWABLE POWER LOSS IN A CIRCUIT

a. The receiver, as you learned earlier in this information sheet, converts electrical power back to sound. It does this by electromagnetic action. The ac voice currents flowing through the receiver windings cause the diaphragm to move in and out at the same frequency. This, in turn, sets up air vibrations producing the sound originally applied at the transmitter.

b. Now the receiver can't produce any sound unless it gets a certain amount of power. This means that there is a limit to how much power can be lost in a circuit and still get sound out of the receiver. The limit for military telephone circuits is a loss of 99.9 percent of the .001 watt (1 mw) sent from the transmitter (fig. 16).

c. This may seem surprising, but it's true. A telephone receiver can produce sound if it receives only .000001 watt (1 microwatt) of voice power. Another way of looking at it is that you can lose 999 parts of the original 1 milliwatt sent from the transmitter and still have enough power left.

d. This, of course, is the bare minimum. It isn't a goal you're trying to shoot for. You want to receive much more power than this 1 microwatt, because the more power received, the louder the sound. The limit you do shoot for is a reduction in loss of 75 percent of the original voice power. When 1 milliwatt of voice power is sent out you try to arrange the circuit so that 1/4 milliwatt (25 percent of the original voice power) is received. It has been found that this amount gives satisfactory sound reproduction. Read these values again so you can remember them.

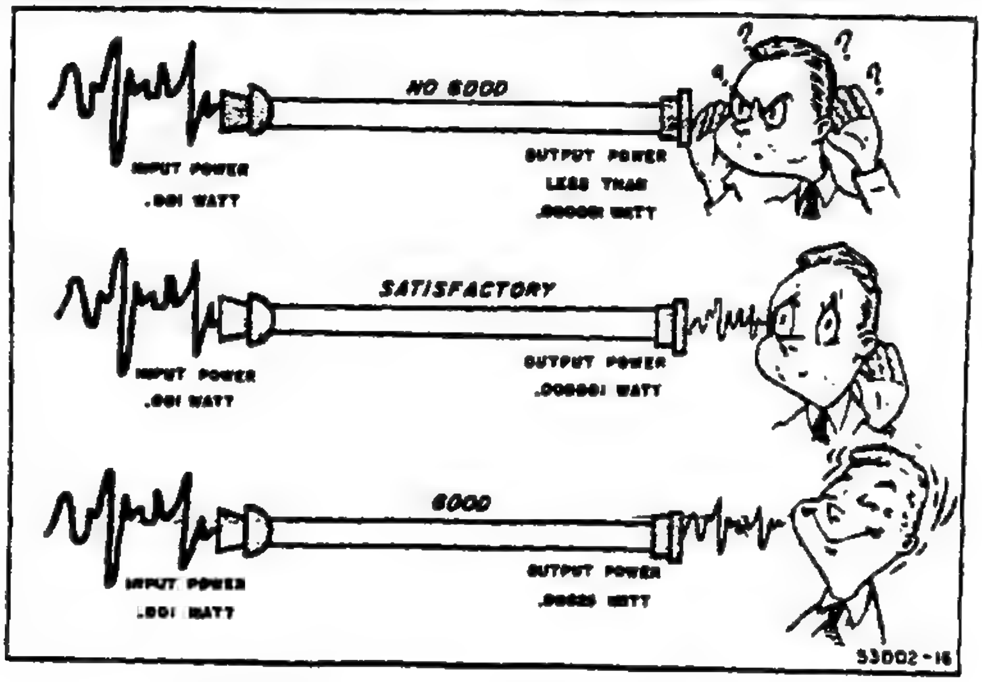


Figure 16. Comparison of voice power received.

(1) Voice power sent from the transmitter is .001 watt (1 milliwatt).

(2) Maximum loss allowable in military circuits is 99.9 percent giving a received power of .000001 watt (1 microwatt).

(3) A reduction in loss of 75 percent of the original voice power is much more desirable since it gives received power of .00025 watt (1/4 milliwatt).

31. SUMMARY

a. You read at the beginning of this sheet that a line limits the distance over which you can transmit. Now you know why this happens. You know that attenuation is what causes this limit. And you realize that attenuation and unequal attenuation are the big problems in long lines transmission.

b. As you go on, you'll learn about other transmission problems. But, to understand them, you must remember what you've learned here. So let's briefly review the main points once more.

(1) A telephone transmission system consists of three main parts:

(a) A source of energy which is the transmitter circuit.

(b) A transmission medium which is the telephone line.

(c) A receiving device which is the receiver circuit.

(2) The transmitter circuit changes sounds of many different frequencies to electrical energy having the same frequencies.

(3) This electrical energy is called voice power and the standard unit for transmitted voice power is .001 watt (1 milliwatt).

(4) Voice power is carried by the telephone line from the transmitter to the receiver.

(5) The telephone line reduces (attenuates) the voice power which it carries.

(6) This attenuation is caused by the line properties, Resistance (R), Leakage (G), Inductance (L), and Capacitance (C).

(7) The line properties cause attenuation which increases as the frequency rises (unequal attenuation). This means that, as the frequency rises, increasingly less voice power reaches the receiver.

(8) The receiver takes the voice power from the line and converts it back to sound.

APPENDIX B

OVERCOMING TRANSMISSION LINE ATTENUATION

IT 53004B

TABLE OF CONTENTS

Note: **This information text supersedes SSTS 53004A, Overcoming Transmission Line Attenuation.**

1. INTRODUCTORY INFORMATION

a. There are several characteristics of transmission lines that cause a signal to be attenuated. However, the signal in a long transmission line may not be completely lost because we can either overcome or reduce the affects of attenuation. This text covers two methods of combatting attenuation -- using repeaters and using loading coils. The difference between the two methods is shown in figure 1. A repeater overcomes attenuation at the repeater site because it amplifies the signal. A loading coil reduces the effect of the shunt capacitance of the line. By reducing the shunt capacitance, attenuation is also reduced and more signal power can get through.

b. Signal power loss is primarily the result of line (I^2R) loss. To reduce the I^2R loss you can either reduce line resistance (R) or line current (I). To reduce the line resistance would require the use of thicker wire which is both impractical and very expensive. To reduce the line current (I), you would have to increase the impedance of the transmission line.

2. COMBINING INDUCTANCE AND CAPACITANCE IN A LINE

a. Inductive reactance (X_L) and capacitive reactance (Xc) have an opposing and cancelling effect upon each other. When X_L and Xc are equal in value, they completely cancel each other. According to the equation $Z = \sqrt{R^2 + (X_L - X_C)^2}$, when $X_L = X_C$ the impedance becomes resistive (Z = R) and is reduced. This condition is called resonance and is usually desirable in a circuit.

b. A telephone transmission line is one of the circuits where resonance is not desirable because it may cause howling or singing at the resonant frequency points. However, for the purpose of power transmission, it is desirable that X_L cancel out most of the Xc.

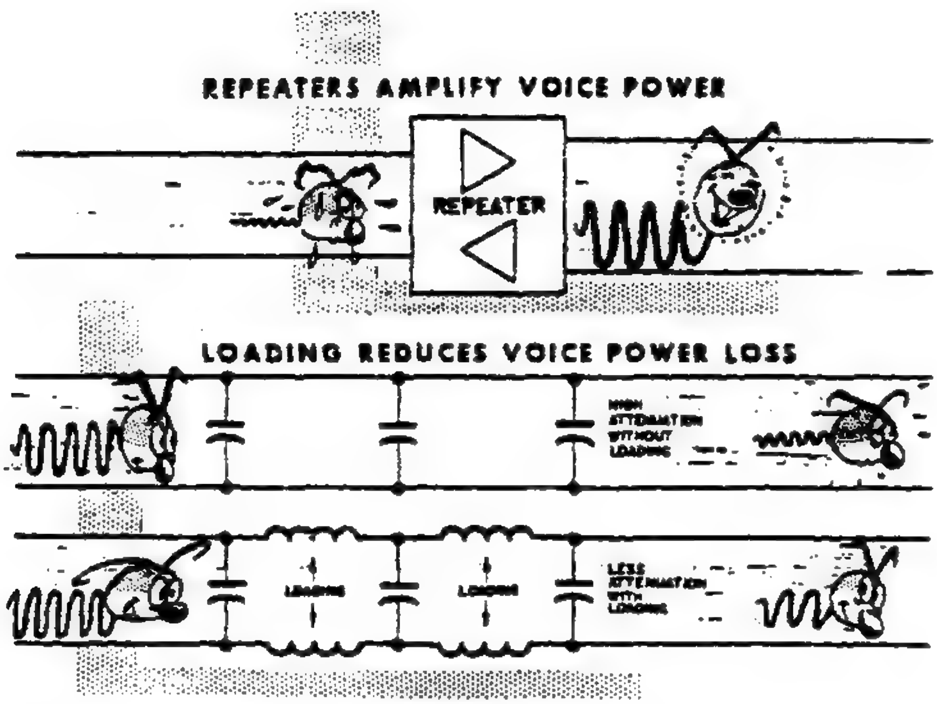


Figure 1. Repeaters and loading work differently.

c. To determine the impedance of a telephone line we can use the formula:

$$Z = \sqrt{\frac{R + jwL}{G + jwC}}$$

Although the formula may look somewhat complicated, we can use it to show the result of adding inductance in a transmission line. Any inductance (L) that we add will also include a resistance (R) factor. Both of these quantities are in the numerator, under the square root sign. If you increase the numerator of the fraction (R and L) without increasing the denominator (capacitance, C), the resultant impedance, Z, will be increased. This means that when you add inductance to a transmission line, you reduce the effective capacitance and at the same time increase the impedance of the line.

d. The increased impedance automatically reduces the line current and this reduces the I^2R loss of the line. The reduction in line current also reduces the possibility of, or prevents, singing or howling. Because the current is reduced, a higher voltage is required to transmit the signal power. Thus, adding inductance to a transmission line produces a "transformer effect" on the line -- the signal is sent out with high voltage, low current, and has low I^2R loss or attenuation.

3. LOADING MEANS ADDING INDUCTANCE

We say that we load a line when we add inductance to it. To add inductance, loading coils are connected in series with a line as shown in figure 2. Two transmission lines are shown in figure 3. One transmission line is loaded and the other one is not. At the left is a graph of the voice frequency input to the line. Notice that all of the voice frequencies have the same power level at the input to the line. At the right of each line is a graph showing how these voice frequencies have been attenuated by the transmission line. The graphs show that the output of the loaded line has less attenuation than the output of the nonloaded line. This means that the voice frequency signals can be sent over a greater distance on loaded transmission lines.

4. LOADING ALSO REDUCES DISTORTION

a. Look at the graph for the nonloaded line (fig 3) and you can see that attenuation increases with a rise in frequency. The higher frequencies are attenuated more than the lower frequencies which is a form of distortion. The distortion is caused by the effective capacitance of the line. This happens because the capacitive reactance decreases as the frequency increases; more high frequency current is shunted from

Figure 2. How loading coils are connected in a line.

one wire to the other and less of it gets to the receiver.

b. Loading reduces the capacitive effect. The graph at the end of the loaded line (fig 3) shows that the attenuation is relatively flat over a large portion of the frequency range. Thus, the voice currents transmitted over a loaded line have less distortion and the voices from the telephone receiver have a more natural sound.

5. LOADING A TRANSMISSION LINE

The effective capacitance of a transmission line is distributed along the entire line; it is not lumped at any one point. It's as if there were thousands of small capacitors connected in parallel along the entire length of the line (fig 4). The distributed capacitance has a shunting effect on voice currents all along the line. Since inductive loading is used to combat the capacitive shunting effect, the loading coils must also be distributed along the entire line. If only one large loading coil were connected in the center, at the end, or anywhere else along the line, it would cause a condition called overloading. An overloaded line causes more attenuation and more distortion than a line without any loading at all. The reason is that the large inductance in a single

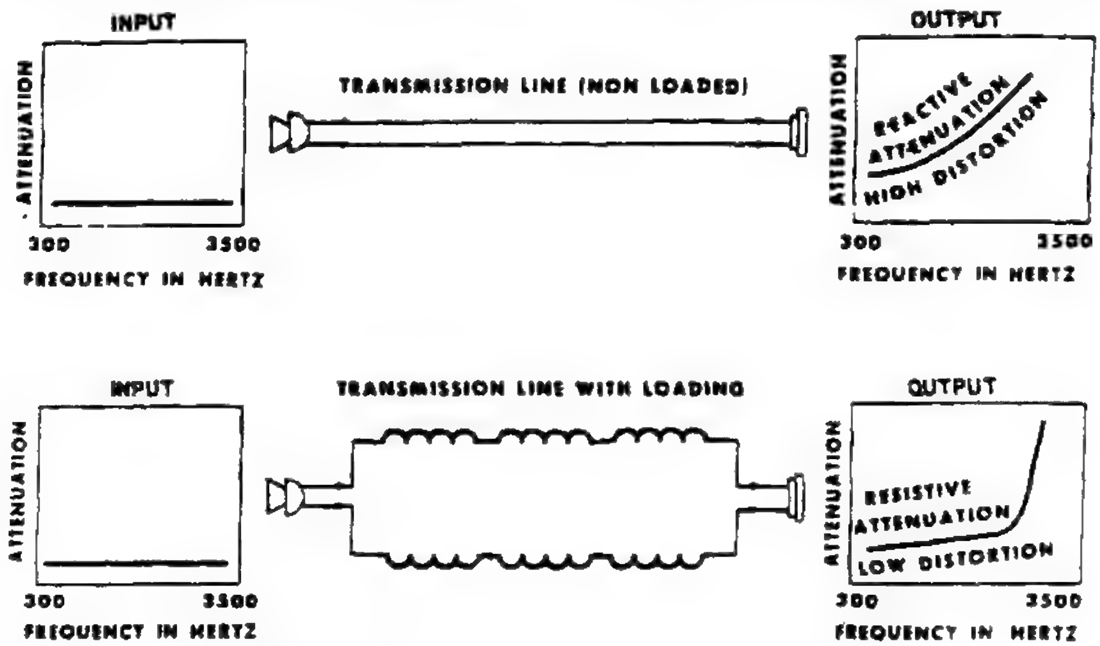


Figure 3. Loading reduces attenuation and distortion.

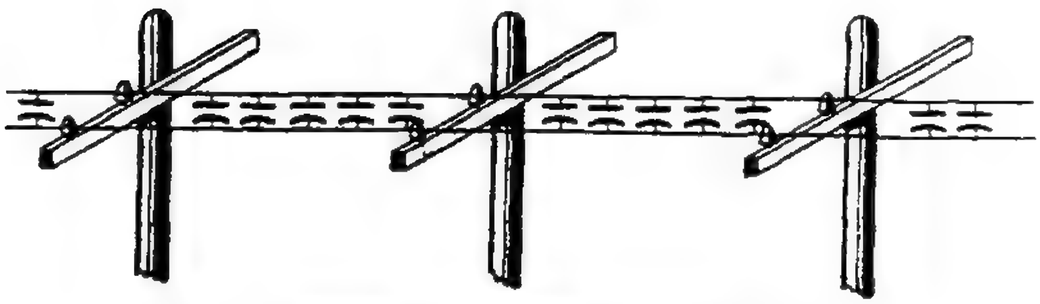


Figure 4. Capacitance is distributed along the entire line.

part of the line cannot counteract the capacitive effect distributed along the entire line. To prevent overloading and to counteract the distributed capacitance, two methods of loading are used.

a. Continuous Loading. Continuous loading consists of wrapping each wire of the pair to be loaded with a magnetic tape (fig 5). The tape, because it is magnetic, causes an inductive field to build up around each of the wires when current is flowing, counteracting the capacitive effect. Continuous loading is the best type of loading that can be used. But, because of its extremely high cost, it is impractical for most transmission lines and is used only on submarine (underwater) cables.

b. Loading Coils. Another method of loading is to insert small loading coils at frequent intervals along the length of the transmission line. The spacing must be such that there are several coils per wavelength of signal; otherwise the attenuation will increase rather than decrease. Loading coils are cheaper than continuous loading and this method is more frequently used.

6. LIMITATIONS OF THE LOADING METHOD

a. Although loading improves the transmission characteristics of a line, the method has limitations. It can only improve transmission to a certain degree; it only reduces line losses and distortion. And there is a limit to the number of loading coils that can be added in a transmission line circuit; too many coils cause loss of signal power and distortion.

b. The combination of inductance and capacitance causes the transmission line to act like a filter. Attenuation increases sharply when the frequency exceeds a certain value; this frequency is called the cutoff frequency. The cutoff frequency for a circuit depends upon the loading coils selected. Notice in figure 6 that the frequencies in the

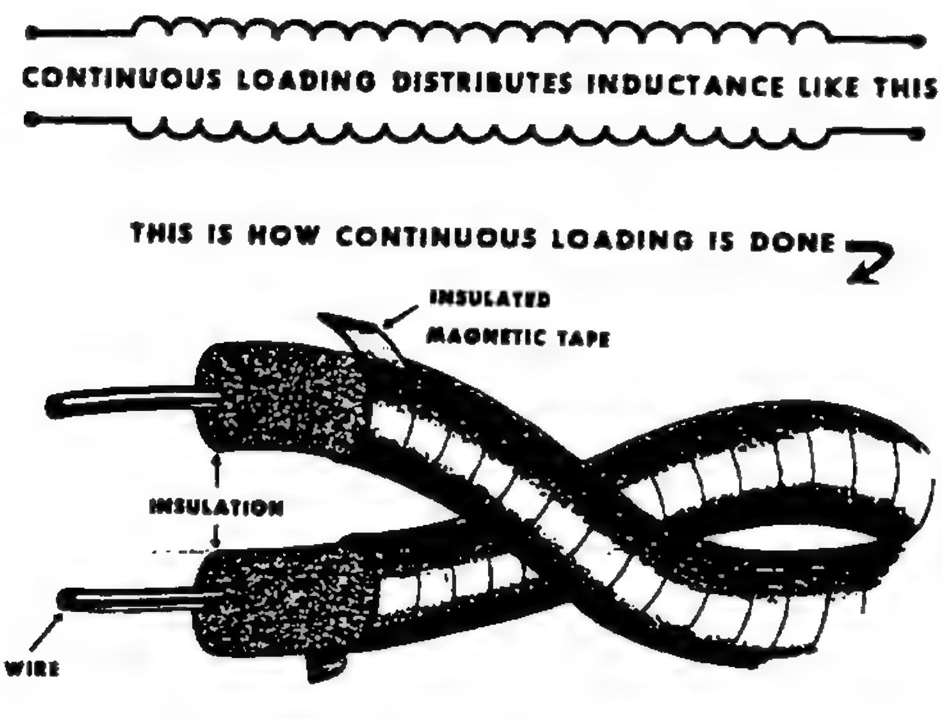


Figure 5. Continuous loading.

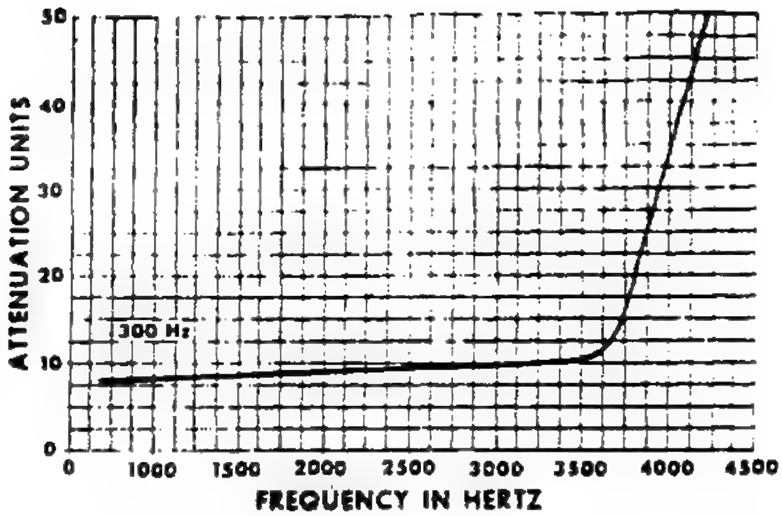


Figure 6. Loading affects frequencies transmitted.

300-3,500 Hertz range get about the same attenuation by the line. Above 3,500 Hertz however, the attenuation increases very sharply. The graph represents the response of a circuit designed to pass only voice frequencies.

7. SPIRAL-4 CABLE

a. Spiral-4 cable is used in many transmission line circuits. It has two pairs of wires (fig 7), is insulated with polyethylene, and is provided in standard 1/4-mile lengths. Spiral-4 cable may be used as a loaded or nonloaded line depending on the transmission line circuit requirements.

b. Each spiral-4 cable has a universal connecter at each end. To load a spiral-4 line, 6 millihenry coils are inserted at 1/4-mile intervals, between cable lengths. The cutoff frequency for spiral-4 cable is about 28,000 Hertz.

8. REPEATERS

a. The main job of a repeater is to amplify (increase) voice power. Repeaters are installed at various points along a line. When attenuated voice power passes through a repeater it is amplified; it gains extra power from the repeater.

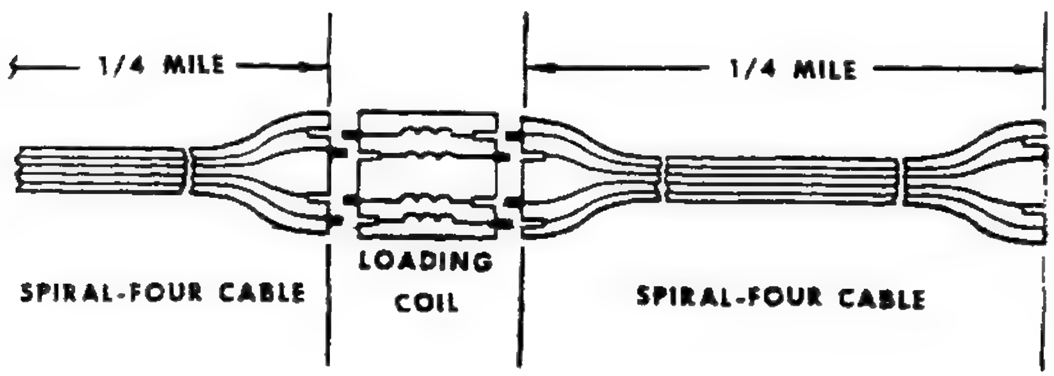


Figure 7. Loading in spiral-four cable.

b. You can see how this works in figure 8. Two transmission lines built from the same type and length of line are shown. A repeater has been installed in one of the lines. Notice that the input power to each line is the same (1 milliwatt). The output power, however, is different for each line. The power has been so greatly reduced in the line that has no repeater that no sound is produced in the receiver. In the other line, the power has been reduced the same amount. Yet at the receiver there is plenty of power left to produce audible sound. The repeater makes the difference. Although both lines give the same attenuation, the repeater has increased the power in one line and has overcome the attenuation.

c. Figure 8 also shows that the section of line extending from the transmitter to the repeater has reduced voice power from 1 milliwatt to 1/1,000 milliwatt by the time it enters the repeater. Inside, amplification takes place and the voice power comes out as 4 milliwatts. If you figured it out, you'd find that the repeater amplified the 1/1,000 milliwatt input 4,000 times. This is a tremendous gain of power and the part of a repeater that does this job is called an amplifier.

9. REPEATERS HAVE TWO AMPLIFIERS

Modern day repeater amplifiers may use electron tubes or transistors. The amplifiers are one-way devices which means that voice power can go through an amplifier in only one direction. Thus, each repeater requires two amplifiers: one for each direction of transmission. Figure 9 shows the symbol for a repeater. The amplifiers are represented by the arrow heads and the arrow heads point in the direction of transmission. The gain provided by the amplifier must overcome the loss caused by all of the other parts of the transmission system.

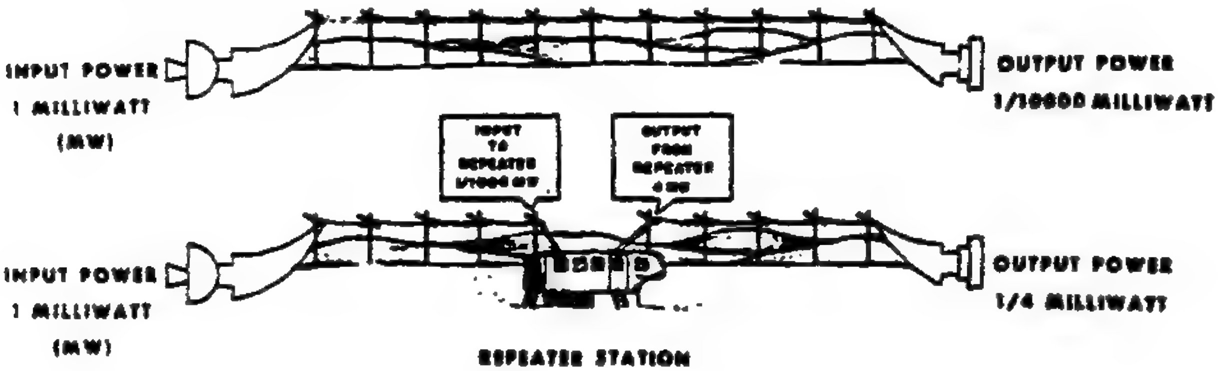


Figure 8. Repeater provides a gain.

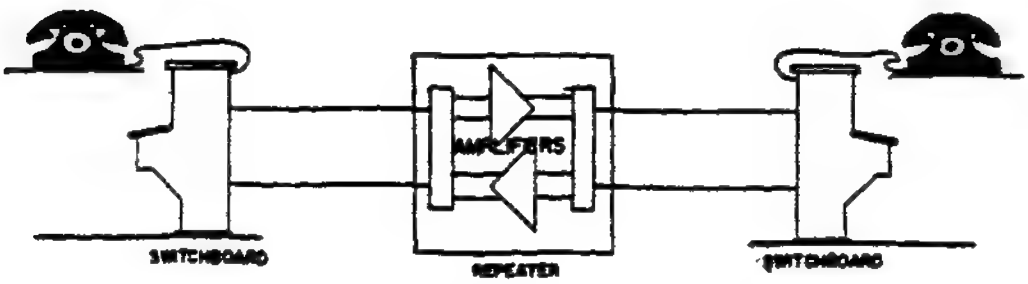


Figure 9. Repeaters have two amplifiers.

APPENDIX C

INTRODUCTION TO
COMPANDORS
SSTS 53105

OBJECTIVES:

1. To discuss principles of sound energy related to compandor operation.

2. To explain what a compandor is.

3. To show what a compandor does.

4. To explain how a typical volume compandor works.

INTRODUCTORY INFORMATION

In voice communications, interfering disturbances called noise and crosstalk constantly try to reduce intelligence to meaningless jumble. A special device which offers a practical method of dealing with these two main enemies of communication is the compandor.

In its important role, the compandor serves as a remedial device designed to transmit voice signals above the noise and crosstalk encountered in communication systems, thereby improving the quality of voice transmission. Refer to figure 1 on next page.

To understand how compandors offer relief from the disturbing effects of noise and crosstalk, you have to know something about how compandors work and about the basic principles of sound energy related to their operation. The purpose of this sheet, therefore, is to provide you with this information.

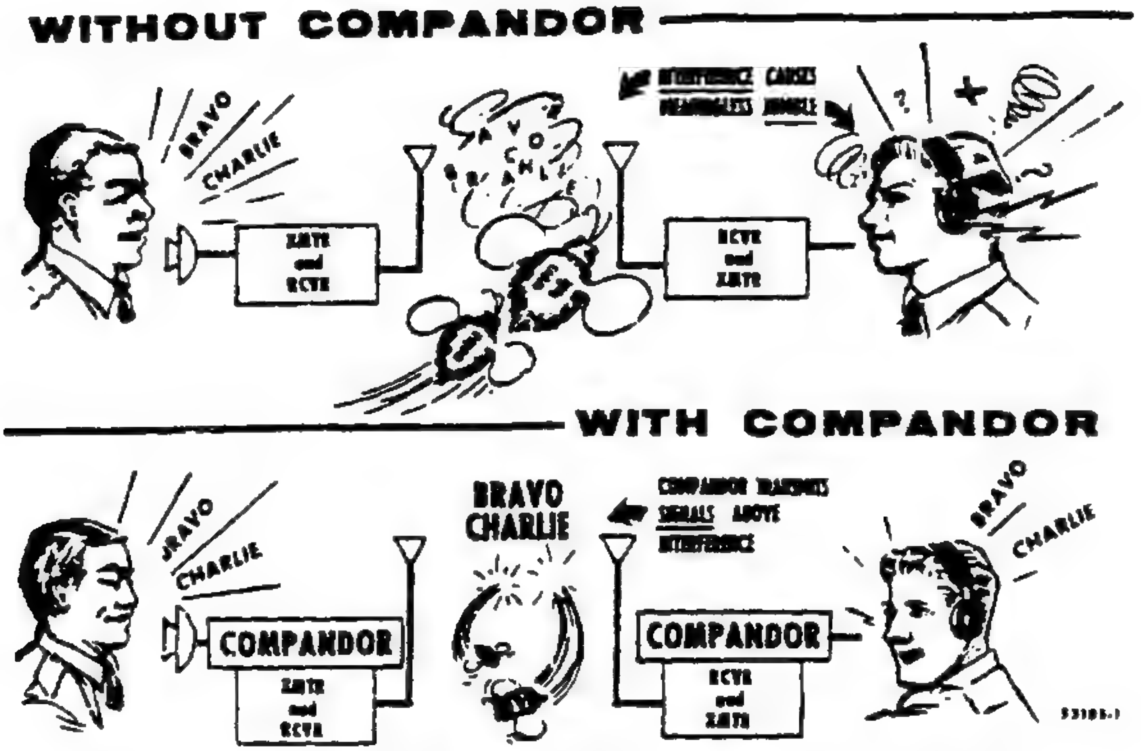


Figure 1. A compandor improves the quality of voice transmission.

Before we discuss compandors in this sheet, we'll first review some basic principles of sound (speech) energy related to compandor operation. Next we'll discuss the two main enemies of communication -- noise and crosstalk. Then we'll explain what a compandor is and show how it offers relief from noise and crosstalk. Next, we'll briefly discuss the different types of compandors and explain how a typical volume compandor works. Finally, we'll tell in what types of equipment compandors are used.

SOUND (SPEECH) ENERGY

In SSTS 53001, Basic Principles Of Sound-Review, you learned that your voice produces tones consisting of different frequencies and varying intensity. And you've learned that together these two signal characteristics, frequency and intensity, make up the speech energy that we convert into electrical energy by means of radio, telephone, or other communications systems. Because these characteristics of speech energy play an important part in the operation of compandors, let's review some other important facts about intensity and frequency.

FIRST, SPEECH FREQUENCY

Speech frequency is the number of vibrations (Hertz) that make up sound. The frequency range (Fig. 2) of human voice (male and female together) is from about 100 to 8,000 Hz. However, most of the energy of speech signals being transmitted is concentrated in a frequency range of 200 Hz to 3 200 Hz.

Remember, that in communication systems, we're not interested in transmitting all our vocal tones. We're interested in transmitting only enough frequencies so that a listener can understand what we say. (For additional information on frequency, review SSTS 53001.)

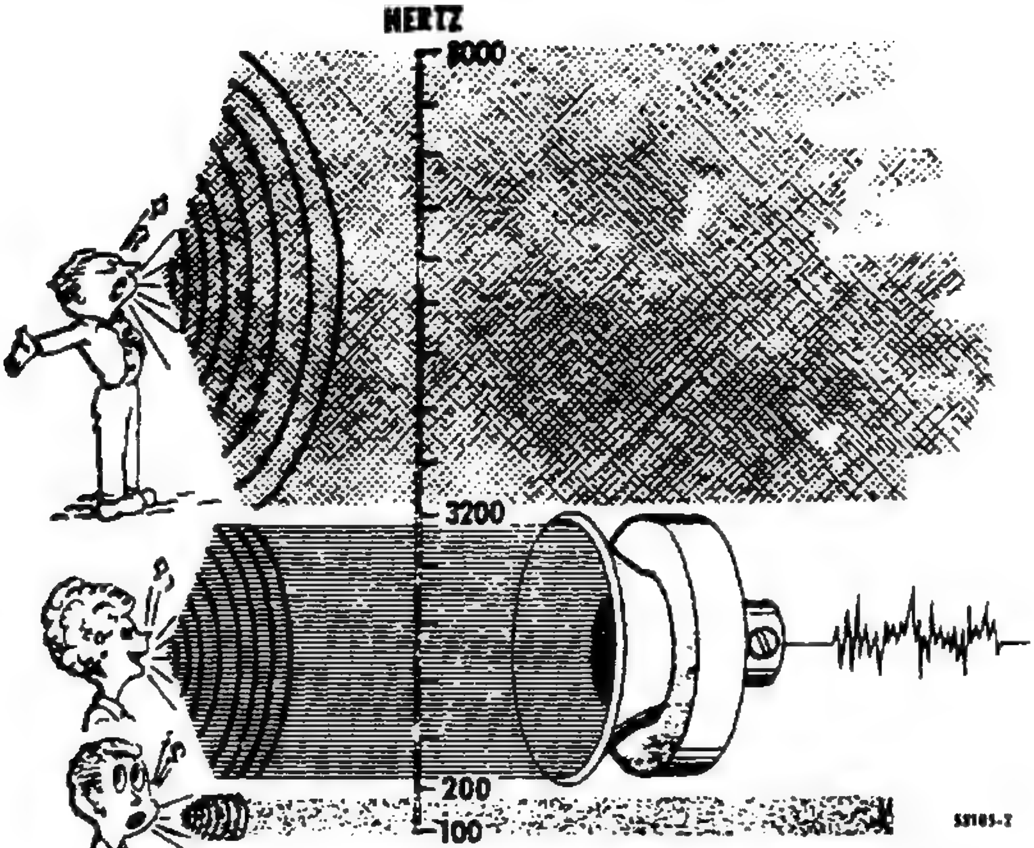


Figure 2. We only transmit part of the voice frequency range.

NEXT, SPEECH INTENSITY

Speech intensity refers to the volume, loudness, or power of speech. A common unit of measurement of intensity is the decibel (db).

The intensity or power level of our speech changes when we speak. Notice in A of figure 3 how this person's speech changes in power level. At point one, the power level is 10 db; at point two, it is 25 db; and so forth. At point four, the speaker reaches his highest power level of 50 db.

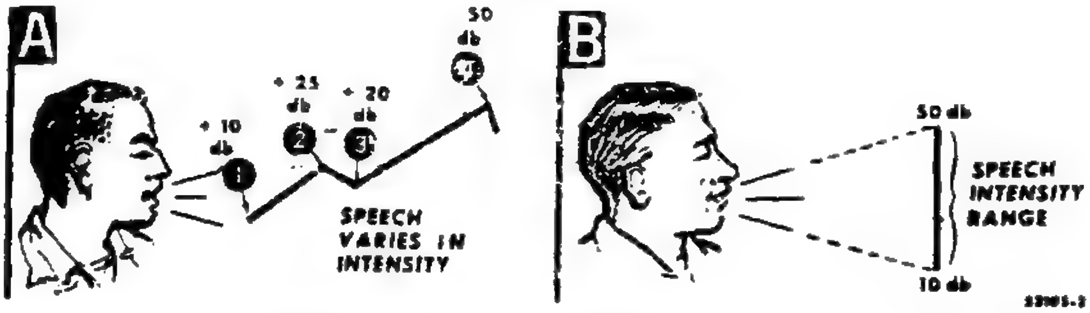


Figure 3. Our speech varies in intensity.

The difference between the speaker's lowest and highest power level shown in part B is 40 db (50 db - 10 db = 40 db). We call this difference the speech intensity range. (Refer to SSTS 53005, Power Losses and Gains -- Decibels, for complete information on db's.)

WHAT DETERMINES SPEECH INTENSITY RANGE?

Two main factors determine speech intensity range: the speaker and his words. Here's why.

A speaker can increase or decrease his speech intensity range by varying his tone of voice. For example, if he changes his voice from a whisper, when talking about "sweet nothings", to loud sounds when stressing a point, his intensity range will be greater than it will be if he speaks in the same tone.

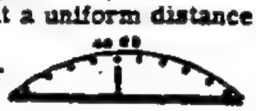
Words also help to determine speech intensity range because our speech syllables produce different amounts of power. What is a syllable? Briefly, a syllable is one or more letters of a word taken together to form a sound. For example, the word America has four syllables (A-mer-i-ca). Each spoken syllable produces a different amount of power (see figure 4).

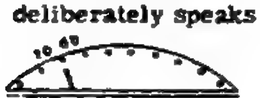
Figure 4. Words help determine intensity range.

Notice in figure 4 that the syllable "mer" produces the most power because the person stresses that syllable more than the others. Thus, "mer" helps to determine the power in the high end of the intensity range. Syllable "A" produces less power because it was stressed less. Therefore, "A" helps to determine the power in the lower end of the intensity range. Of course, the amount of power that syllables in this or any other word produce depends on the way the speaker stresses each syllable.

The normal intensity range for the average person is about 40 db; however, some people have an intensity range of 60 db or more.

TRANSMITTING A WIDE RANGE OF SPEECH POWERS

In communications, an operator's successive speech signals may differ in intensity by many decibels even though he may have an average intensity range. For example, when an operator speaks evenly at a uniform distance from the microphone, he establishes an average intensity of speech power  But if the operator deliberately shouts, stresses certain words or syllables, or moves closer to the microphone, the intensity increases . However, if the operator moves farther away from the microphone, or deliberately speaks in a quiet tone of voice, the intensity of speech power decreases.



Notice that the difference between the operator's loudest and weakest sounds recorded on the meters (page 4) is 60 db (80db-20db). This means that there is 1 million times as much power in the loudest sound as there is in the weakest sound because 60 db is equivalent to a power ratio of 1 million to 1. Transmitting such a wide range of speech powers presents some problems in communication systems.

PROBLEMS IN TRANSMITTING WIDE RANGE OF SPEECH POWERS

Two major problems encountered in transmitting wide ranges of speech power are as follows:

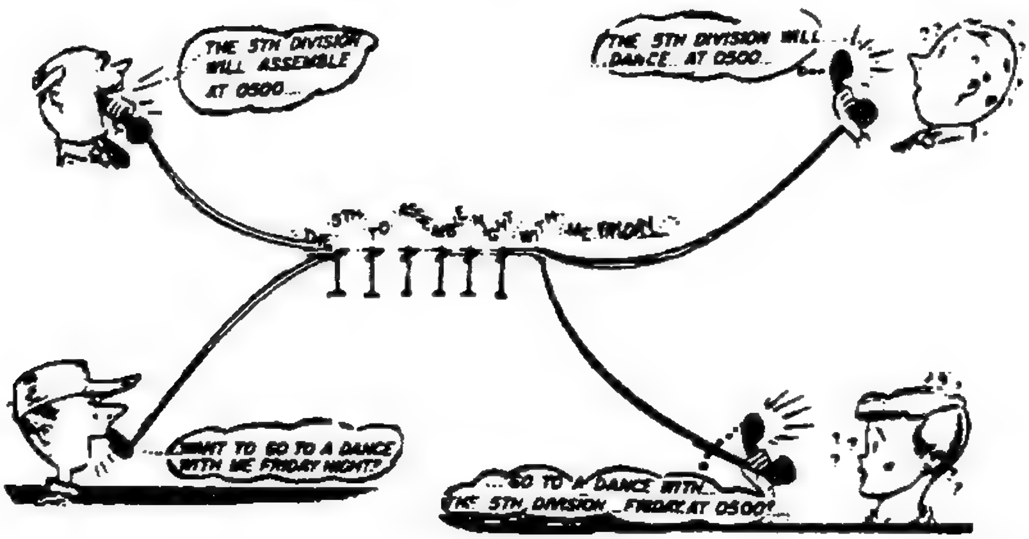
1. Low power (weak) signals become lost in electrical noise and crosstalk -- the two main enemies in communication systems.

2. High power (strong) signals constitute a major source for crosstalk.

Now what is crosstalk and noise? Let's find out by discussing each of these disorganizing forces separately.

FIRST, CROSSTALK

Crosstalk is interference (unwanted signals) that leaks from one channel to another in a communication system. When conductors that are close to each other carry messages as shown in figure 5, some signal power slips from one conductor to another, causing channel interference with the desired conversation. All of us have been exposed to crosstalk at one time or another while talking on the phone, listening to the radio, or watching TV.



53106-5

Figure 5. Crosstalk interferes with conversation.

Crosstalk may invade the privacy of a communication system at any time. But this type of interference is most noticeable and annoying during silent periods of a conversation, such as at the end of a sentence, between words, or during any other breaks in speech.

NOW, ELECTRICAL NOISE

Electrical noise is by far the principal enemy of communication and is the type of interference you'll be concerned with most in your work as a repairman. This type of interference is much more serious than crosstalk because it can come from any one of a number of sources. Besides, noise is generated afresh at almost every point in the transmission path. Now let's learn about electrical noise by answering these three questions:

1. What does electrical noise look like?
2. Where does electrical noise come from?
3. Why is electrical noise harmful to communications?

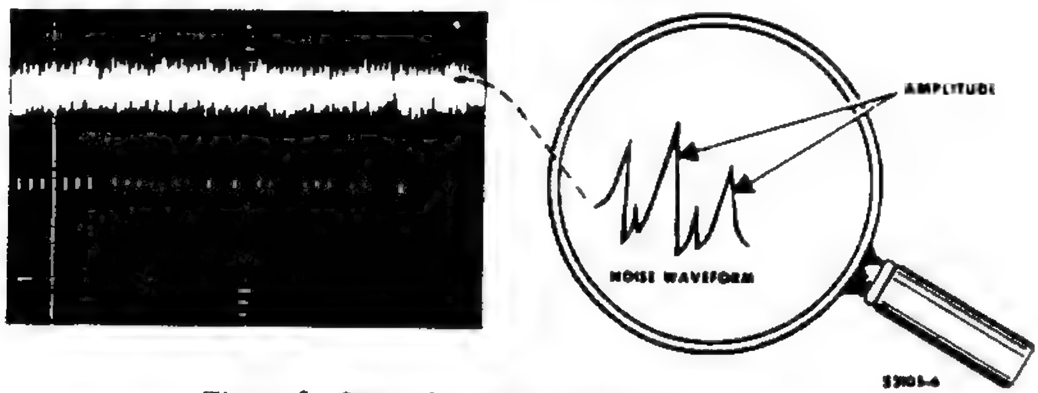


Figure 6. What electrical noise looks like.

FIRST, WHAT DOES ELECTRICAL NOISE LOOK LIKE?

Electrical noise takes on many shapes and forms. Figure 6 shows how electrical noise in communication channels may appear on an oscilloscope. Note that there is no particular pattern to the noise. The amplitude of the noise keeps changing; power level and frequency keep changing also.

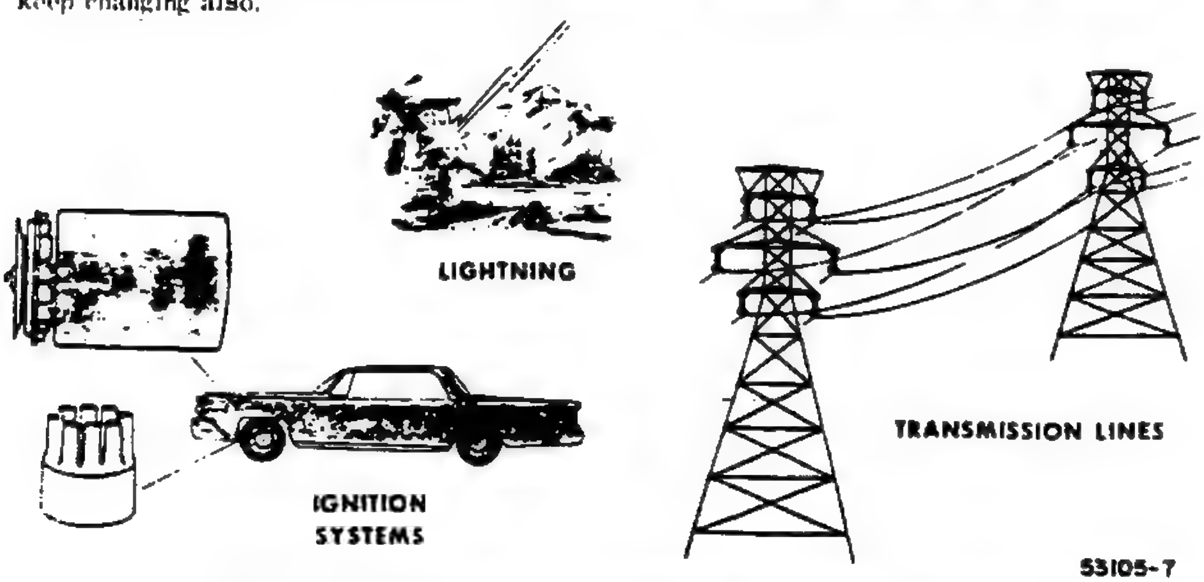


Figure 7. Sources of electrical noise interference from outside communication systems.

NEXT, WHERE DOES ELECTRICAL NOISE COME FROM?

Electrical noise interference comes from both outside and inside the communication system, however, most noise comes from outside the system. Outside noises originate from such sources as lightning, electrical-power transmission lines, and automobile ignition systems (see figure 7). These and similar sources radiate noise, in the form of electrical energy, in all directions; nearby communication systems pick up the radiated noise.

Noise within the communication system is generated by internal components of transmitting and receiving devices. You see, the flow of electricity through components causes electrons to bump into molecules within conducting material and, thus, set up noise. Notice that the movement of electrons in the resistor, shown in figure 8, generates a small electrical current (noise). You can also detect similar noise generated in tubes, transistors, and other components found within electronic equipment.

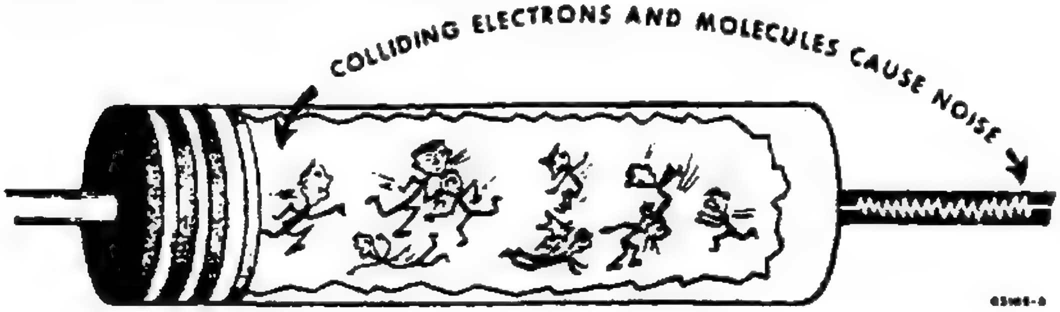


Figure 8. Where electrical noise interference from inside communications systems comes from.

FINALLY, WHY IS ELECTRICAL NOISE HARMFUL TO COMMUNICATIONS

The presence of noise in a communication system, regardless of where comes from, makes it difficult or impossible to hear the entire range of an operator's voice. You can hear the loudest parts of his voice because they are more powerful than the noise, but you cannot hear the quietest parts because of "masking." That is, the noise drowns out the weaker signals (see figure 9).

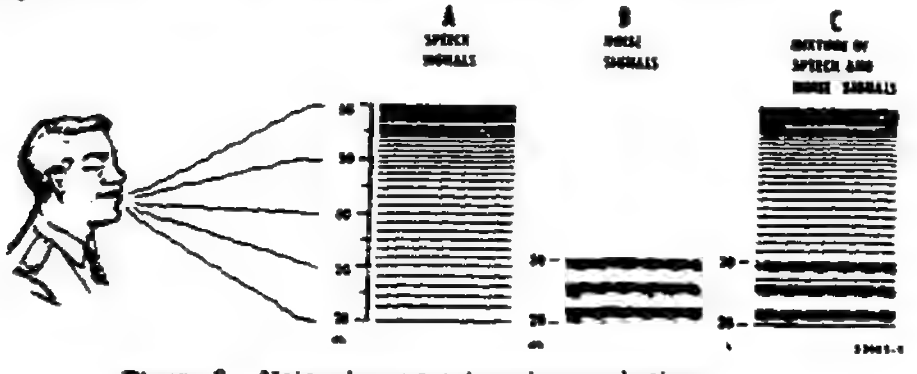


Figure 9. Noise drowns out weak speech signa .

Notice in A of figure 9 that the operator's speech signals extend over an intensity range of 40 db (60db-20db). Also notice in B of figure 9 that the assumed noise level in a communication system is 30 db. Now when the speech signals combine with the noise signals (C of figure 9), you'll notice that the noise level is sufficient to mask the operator's weak speech signals (signals below 30 db), thus drowning out parts of the operator's conversation.

Let's look at the "masking" effect that noise has on speech signals in another way.

ANOTHER LOOK AT THE WAY NOISE AFFECTS SPEECH SIGNALS

Another way to see how noise drowns out weak speech signals is to look at the shape of a speech signal before and after it has been exposed to the disorganizing force of noise (see figure 10).

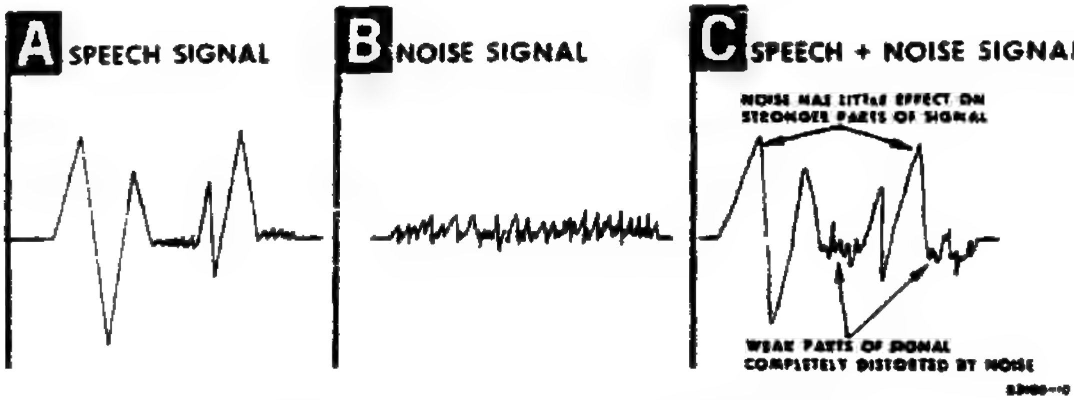


Figure 10. Noise distorts weak speech signals.

Refer to the shape of the speech and noise signals in parts A and B of figure 10. Now, look at part C and notice the shape of the speech signal after it mixes with the noise signal. Observe that the noise signal distorts the weaker parts of the speech signal. The reason -- the noise signal is more powerful than the weaker parts of the speech signal. In an actual communication system, the operator would not hear the weaker parts of this signal; instead he would hear noise plus the stronger parts of the signal.

You should also notice in part C that noise attacks the strong as well as weak parts of the speech signal. However, because the strong parts of the signal are so much more powerful than noise, noise has little or no effect on strong signals. Because noise has little effect on strong signals, the remainder of this text will deal with the effect of noise and crosstalk on weak signals and not on strong signals.

WHAT WE CAN DO ABOUT INTERFERENCE FROM NOISE AND CROSSTALK

Now that you've learned what noise and crosstalk are and how they affect communications, what can we do about them? It's impossible to get rid of these interferences entirely because, as you've already learned, they can creep into communication systems at any time. It's possible, however, to reduce their effects by improving the signal-to-noise ratio. Let's see what this means.

IMPROVING THE SIGNAL-TO-NOISE RATIO

The signal-to-noise ratio is a comparison of signal strength to noise strength. This ratio, usually stated in db, tells how much greater the signal power is above the noise power. For example, if there is 10,000 times as much power in a certain speech signal compared to the noise produced ~~in a communication system~~, the signal-to-noise ratio is 10,000 to 1 or 40 db (see A of figure 11). But if there is only 10 times as much power in the signal compared to noise, the signal-to-noise ratio is 10 db (see B of figure 11).

If we assume that the noise level is constant, and the signal-to-noise ratio becomes smaller, then obviously, the power in the speech signal also becomes smaller. If this is allowed to go too far, the weak signals will become lost in the noise (and crosstalk), thus destroying communications.

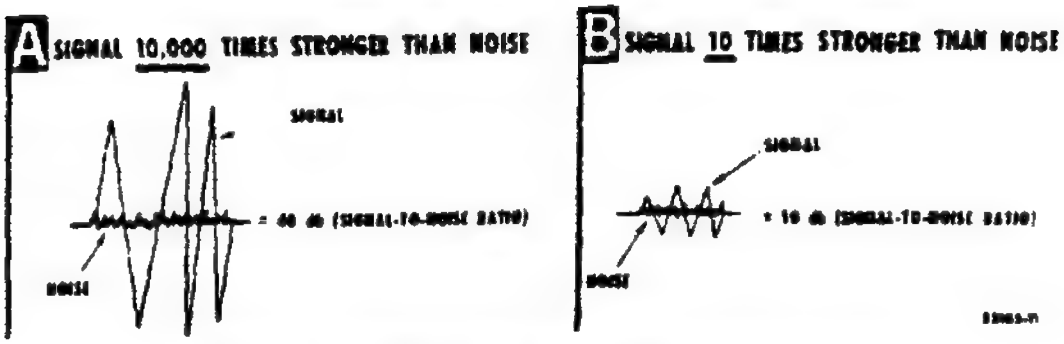


Figure 11. Comparing signal strength to noise strength.

To prevent noise and crosstalk from masking weak signals in a communication system, we must increase the signal-to-noise ratio. That is, we must increase the power of the weak signals above that of the noise and crosstalk. How can we do this? By using a special device called a COMPANDOR.

What is a COMPANDOR? We'll see right after we briefly review the information that we've discussed so far.

REVIEW EXERCISE

Answers to this review exercise are on page 20.

The test items below will help you review the information you have learned about sound (speech) energy, noise, crosstalk, and signal-to-noise ratio. Read each item carefully and place your answer in the space(s) provided.

1. The two signal characteristics that make up speech energy are ____________________ and ____________________.

2. What is the frequency range of most speech signals that are transmitted?

________ ________________.

3. Some other terms for speech intensity are ____________________, ____________________ and ____________________

4. The difference between a speaker's lowest and highest speech power level is called ____.

5. The amount of power that a syllable produces when spoken depends upon ____.

6. List two major problems encountered in transmitting wide ranges of speech power.

____

____

7. The principal interfering disturbance in voice communication is called ___________

8. Briefly -- what is "masking"? ____

____

9. One way to reduce the ill effects from noise and crosstalk is to improve the ________

____

IN GENERAL, WHAT IS A COMPANDOR?

A compandor is an electronic device consisting of two separate voice-operated units. One unit is an intensity range COMpressor; the other is an intensity range exPANDOR. By combining parts of each name we arrive at the name, COM-PANDOR. See figure 12.

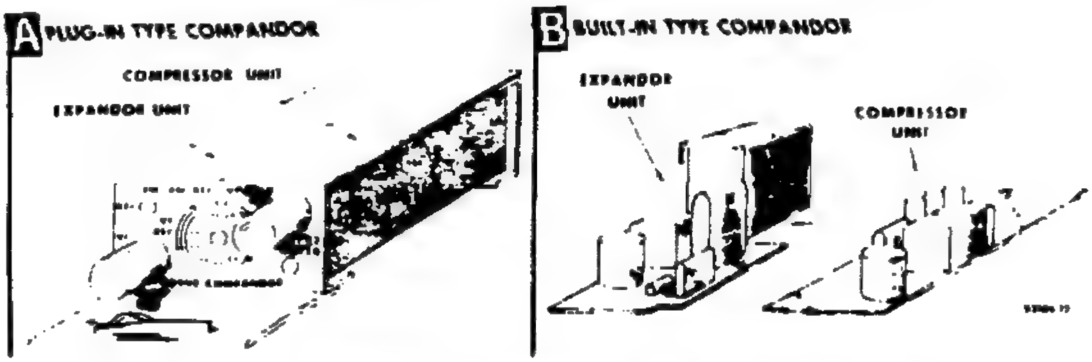


Figure 12. A compandor is a combination intensity range COMpressor and exPANDOR.

In actual equipment, you'll find that the compressor and expandor units may be combined as a single plug-in type unit as shown in A of figure 12, or they can be separate units built permanently in the equipment as shown in B of figure 12. However, regardless of whether the units are together or separate, you should remember that a compandor consists of both a compressor and an expandor.

IN GENERAL, WHAT DOES A COMPANDOR DO?

A compandor improves the quality of voice transmission by decreasing the effects of interference from noise and crosstalk. See figure 13.

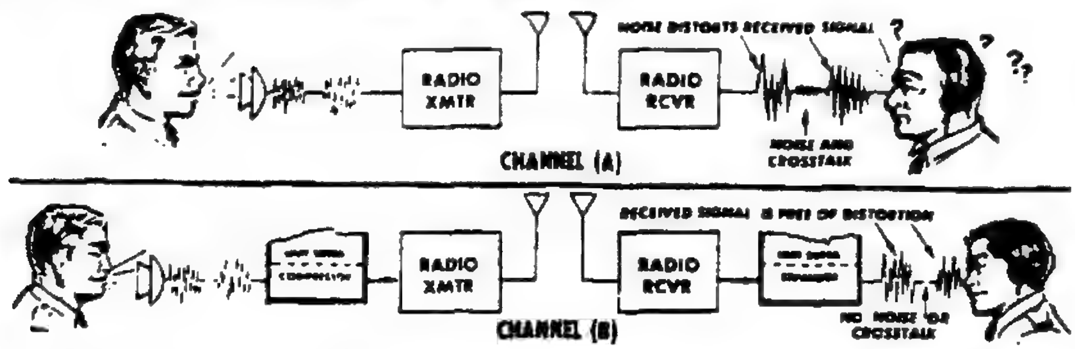


Figure 13. Comparison of a communication channel with and without a compandor.

Notice in channel A of figure 13 that the listener at the receiver end of the channel is confused. The reason -- he isn't getting the complete message because of interference from noise and crosstalk. Notice that not only has noise distorted the received signal, but noise and crosstalk have crept in between words, confusing the listener further.

Now look at channel B of figure 13. Notice that the listener in this channel is not confused: he hears everything the other fellow is saying. Why? Because the compandor (compressor and expandor) decreases the interference resulting from noise and crosstalk. Now the received signal is free from distortion, and just as important there is no noise or crosstalk between the words.

Before we discuss the details of how compandors work, first notice the physical location of the compandor units as shown in channel B of figure 13. Observe that the compressor is located at the transmitting end of the communication channel, while the expandor is located at the receiving end of the same channel. In the equipment that you'll repair, you'll find this the normal arrangement of compandor units. Now, let's discuss how compandors work.

HOW COMPANDORS WORK (GENERAL)

You've learned that compandors are remedial devices offering relief from the effects of noise and crosstalk. But how? To find out, you'll have to study the two units that make up the compandor: the compressor unit and the expandor unit.

FIRST, THE COMPRESSOR UNIT

A compressor unit performs the following two important functions:

1. It raises the power level of weak signals (increases signal-to-noise ratio) so weak signals can be transmitted above the noise and crosstalk encountered in the system between the compressor and expandor.

2. It attenuates (cuts down) very strong signals to help prevent crosstalk from spreading to other communication channels.

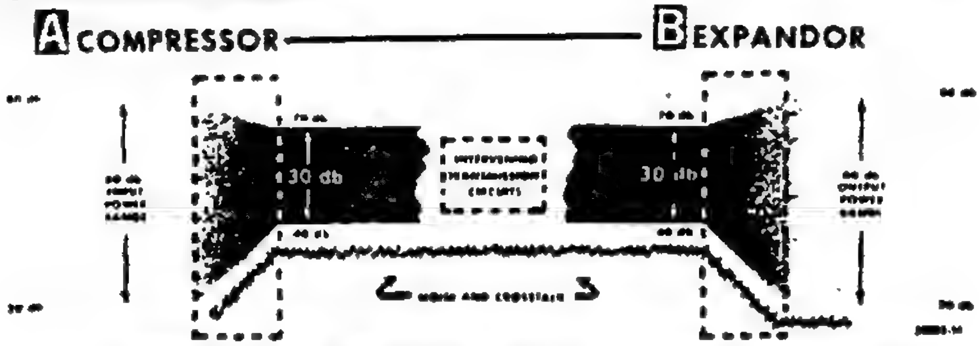


Figure 14. The overall action of a compressor and expandor.

A compressor accomplishes these functions by automatically compressing (reducing) the intensity range of speech signals before transmitting the signals. Notice in A of figure 14 that the input power range to the compressor is 60 db, but the output range is 30 db. This change in range is the result of a shift in the amount of power imparted to the speaker's sounds. Look closely and you will see that the weakest sounds increased in power (from 20 db up to 40 db), while the very strongest sounds decreased in power (from 80 db down to 70 db). In other words, the input power range was compressed by 30 db. Now remember we are transmitting the same sounds but the sounds transmitted are at different power levels from the original power levels. The power that weak signals gain through compression helps them to overcome the noise and crosstalk encountered between the compressor and expandor. The power that the strong signals lose serves to cut down on crosstalk to other channels.

NOW, EXPANDOR UNITS

An expandor unit works opposite to the compressor unit. That is, the expandor restores the reduced power range back to its original range once the speech signals reach their destination. How does the expandor do this? Let's see.

Notice that the power range at the input to the expandor (shown again in figure 15) is 30 db, but the output power range is 60 db. This change in power range is the result of another shift in the amount of power imparted to the speech sounds. However, note that the shift in power is opposite to that shown in the compressor. That is, the weakest sounds decreased in power (from 40 db down to 20 db), while the strongest sounds increased in power (from 70 db up to 80 db). In other words, the expandor restored the 30 db power range, produced by the compressor, back to its original range of 60 db.

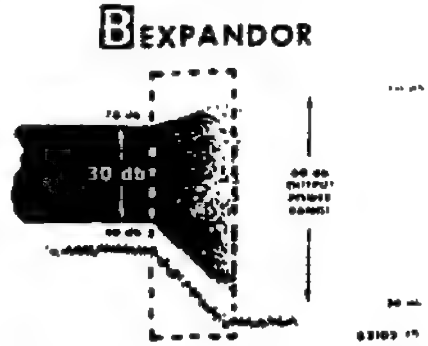


Figure 15. Action of an expandor.

Now how do compandors decrease the effects of noise and crosstalk in communication systems? By simply raising the power level of weak signals (increasing the signal-to-noise ratio) before the signals are exposed to channel interference.

It is important for you to know that a compandor does not eliminate noise and crosstalk, but offers relief from their effects. This is why we call the compandor a remedial device.

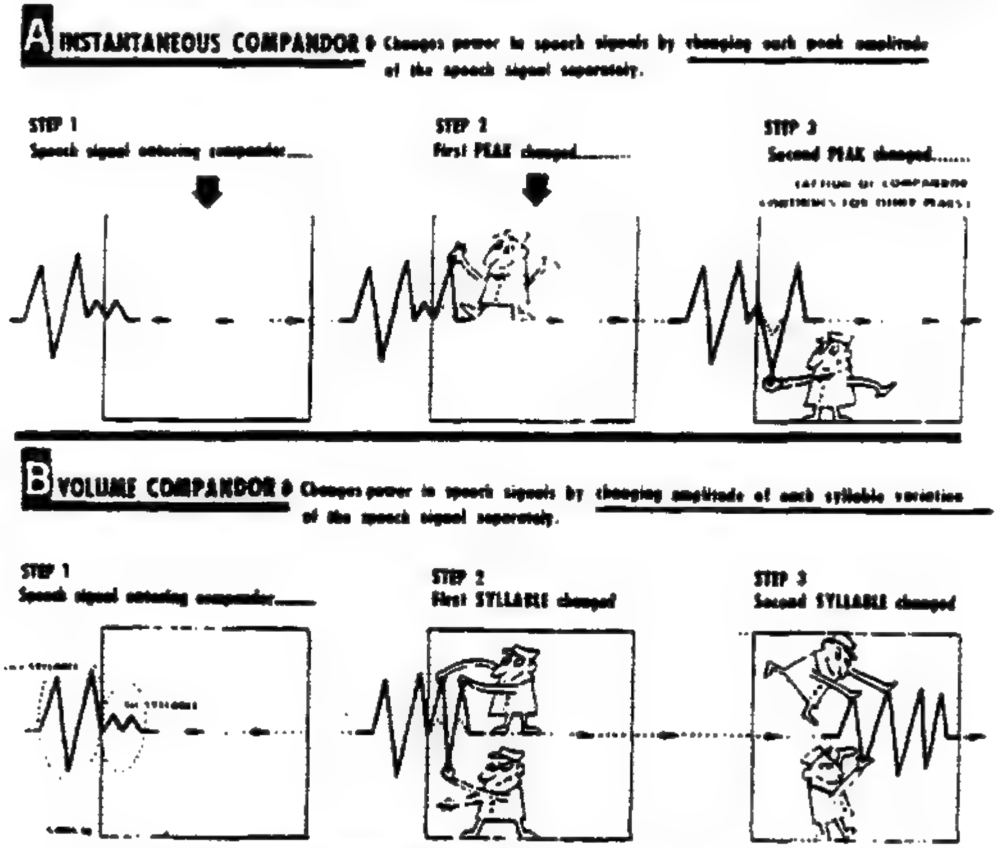


Figure 16. Compandors differ in the way they change power in speech signals.

BASIC TYPES OF COMPANDORS

Basically, there are two main types of compandors: a volume (or syllabic) compandor and an instantaneous compandor. The main difference between the two compandors is the way they change the power of speech signals. To get an idea how the two compandors vary the power of speech signals, study the steps outlined in A and B of figure 16.

In figure 16, notice that the input speech-signal to both compandors is the same; the only difference is the way each compandor changes the power level of the speech signal. Be sure you understand that the instantaneous compandor changes the power by changing each instantaneous peak amplitude of the signal. The volume compandor, however, changes the power in the signal by changing the amplitude of each syllabic variation (rather than each individual peak). Remember that both types of compandors do the same job, but in a different way.

In the remainder of this text, we will discuss only the volume compandor because this is the type used in the transmission system you'll be repairing. The instantaneous compandor is used in pulse-code modulation (PCM) transmission systems.

FIRST, A BLOCK DIAGRAM OF A TYPICAL VOLUME COMPANDOR

The volume compandor, like other compandors, combines both a compressor and an expandor. Each of these units (often called companding units) contains a VARIABLE LOSS DEVICE, AMPLIFIER, and RECTIFIER CONTROL CIRCUIT. Locate these component parts in the compressor and expandor units shown in figure 17.

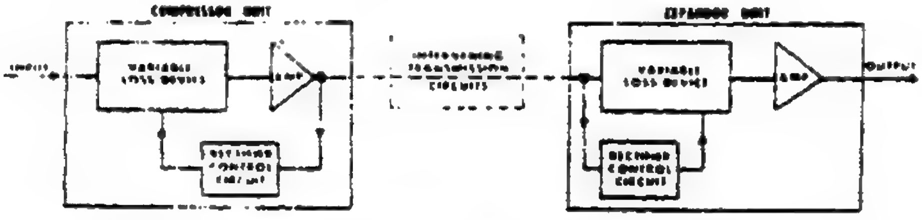


Figure 17. Block diagram of a typical volume compandor.

The three component parts you located in each of the companding units are responsible for changing the intensity range of speech signals. The parts in the compressor work together to decrease the intensity range, while similar parts in the expandor work together to increase the intensity range.

Now that you know the names of the component parts in the companding units, let's briefly discuss the purpose of each part.

1. Variable Loss Device -- determines the overall gain of each syllabic variation of the input signal. In the compressor unit, the variable loss device attenuates strong signals (syllables) more than it does weak signals (syllables); but in the expandor unit, it attenuates weak signals more than it does strong signals.

2. Amplifier -- increases the power of all signals that leave the unit.

3. Rectifier Control Circuit -- controls the amount of signal (syllable) attenuation that takes place in the variable loss device.

You should notice that the component parts in the expandor are connected in reverse to the way the parts are connected in the compressor. The reason -- it is necessary for the expandor to reverse the action of the compressor.

NEXT, HOW A TYPICAL VOLUME COMPANDOR WORKS

To understand specifically how a volume compandor works, you need to know how the component parts you just learned about work together in each companding unit. A good way to learn how the parts work together is to trace the signal path through the compressor unit and then trace the signal path through the expandor unit.

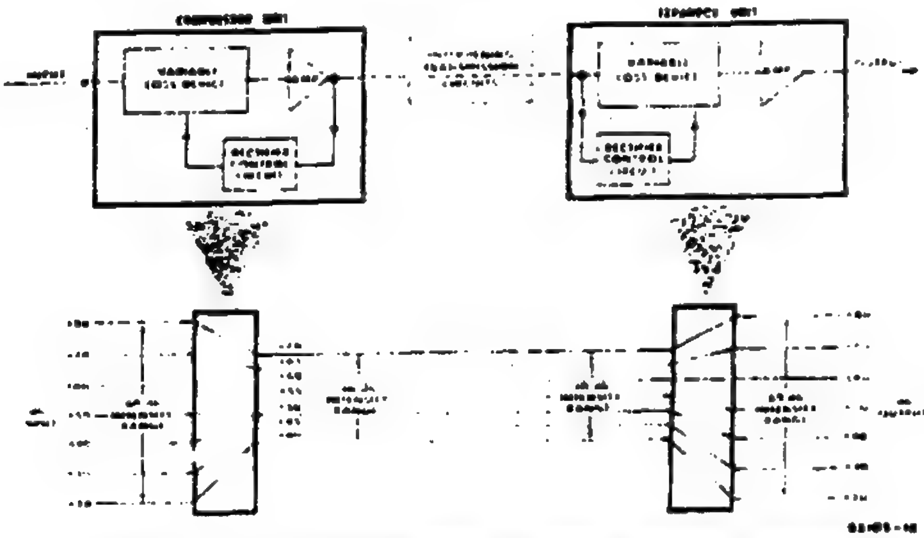


Figure 18. The overall action of a typical volume compandor.

FIRST, THE COMPRESSOR UNIT

As you read, follow the arrows in the compressor unit of figure 18. A speech signal enters the compressor and passes through the variable loss device and then through the amplifier. A part of the output signal from the amplifier goes to the rectifier control circuit where it changes to a control current. In turn, this control current goes to the variable loss device that controls the amount of input signal attenuation. Although all this may sound complicated, you'll see as you read further, that it isn't so complicated after all.

It is important for you to understand that the level of the control current varies directly with the strength of the incoming speech signals. That is, if a weak speech signal is present at the variable loss device, the control current is small and the attenuation of the weak signal is very low (your eye cannot detect the small amount of attenuation). However, if a strong speech signal is present, the control current is high and the attenuation of the signal is high.

In other words, if a signal from a two-syllable word coming into the variable loss device looks like this , when it leaves the device and goes to the amplifier, it will look like this . Notice that the variable loss device attenuates the strong part of the signal more than the weak part. This attenuation of the strong signals results in the compression of the high end of the intensity range.

Now, how does the low end of the intensity range become compressed? This is accomplished by not attenuating the weak signals too much before sending them to the amplifier where they increase in power.

Another name for the compression action, performed by the component parts in the compressor, is companding action.

Refer to figure 18 and note the results of the companding action of the compressor on the different power levels of the 60 db input intensity range. Notice that each input power level below 60 db increases in power while each power level above 60 db decreases in power. Through this companding action, the intensity range is compressed from 60 db to 30 db. This amount of compression is sufficient to overcome the effects of noise and crosstalk that would be encountered between the companding units.

NOW, THE EXPANDOR UNIT

Follow the arrows in the expandor unit of figure 18, as you did in the compressor unit, and remember that the action of the expandor is just opposite to that of the compressor.

A portion of the input signal from the compressor enters the expandor and goes to the rectifier control circuit where it is converted to a control current. In turn, this control current goes to the variable loss device that controls the amount of signal attenuation.

Now, the level of the control current in this unit varies inversely with the strength of the incoming speech signal. That is, if a weak signal  (amplified in the compressor) is present at the variable loss device, the control current is large  and the attenuation of the signal in the variable loss device is high . But, if a strong signal  is present at the variable loss device, the control current is small  and the attenuation of the signal is low . In other words, if a signal from a two-syllable word coming into the variable loss device looks like this  when it leaves the device and goes to the amplifier, it will look like this . Notice that the variable loss device attenuates the weak part of the signal more than the strong part. Attenuating weak signals more than strong signals expands the compressed intensity range back to its original range.

In figure 18 notice the results of the companding action of the expandor on the different power levels of the 30 db intensity range. Each power level above 60 db has increased in power while each power level below 60 db has decreased in power. The net result is that the intensity range has expanded from 30 db back to the natural power range of 60 db.

NOW, CHARACTERISTICS THAT CONTROL THE PERFORMANCE OF VOLUME COMPANDORS

The characteristics that control the performance of a compandor are compression-expansion ratio, companding range, and attack and recovery times. We'll discuss each of these characteristics separately starting with compression-expansion ratio.

FIRST, COMPRESSION-EXPANSION RATIO

A compression and an expansion ratio represents the degree to which speech signals are compressed and expanded. Both ratios are expressed (in db) with figures based on the relationship of input to output power ranges. See figure 19.

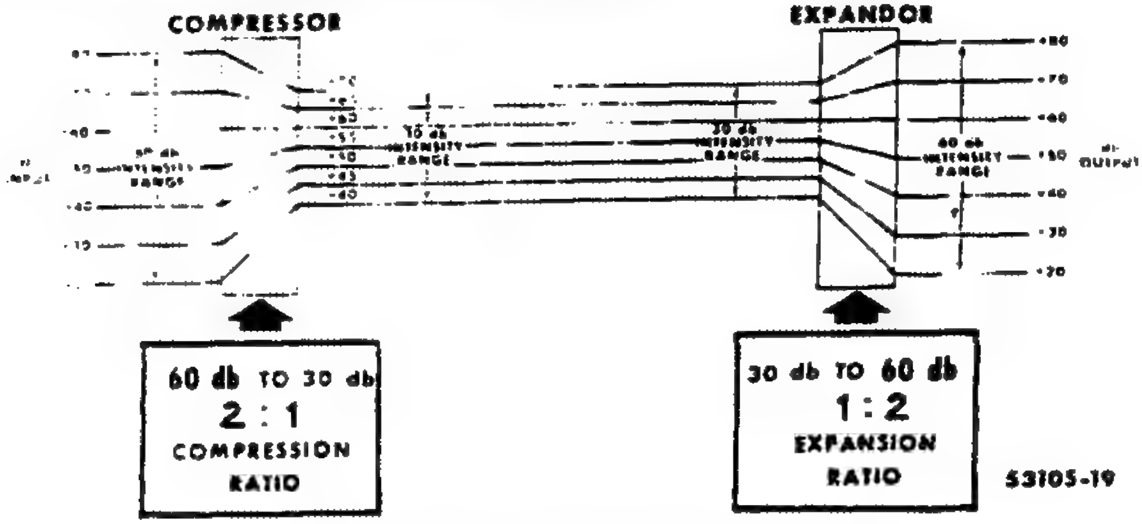


Figure 19. Compression and expansion ratios.

Notice in figure 19 that the input intensity range to the compressor is 60 db, but the output power range is 30 db. The compression ratio, therefore, is 60 db to 30 db or 2:1. A compression ratio of 2:1 means that the speech energy, traveling in the system between the compressor and expandor, has an intensity range of one-half its original value.

Now, look at the expandor unit of figure 19. Notice that the input intensity range to the expandor is 30 db, but the output power range is 60 db. Therefore, the expansion ratio is 30 db to 60 db or 1:2. An expansion ratio of 1:2 means that the intensity range leaving the expandor is twice as great as the intensity range entering the expandor. Thus, the expansion ratio is just the reverse from that of the compression ratio.

You'll find that in most volume compandors, the compression ratio is 2:1 and the expansion ratio is 1:2. The reason -- a compression ratio greater than 2:1 can excessively distort speech signals, while a smaller compression ratio will not sufficiently improve the signal-to-noise ratio for weak signals.

NEXT, THE COMPANDING RANGE

The companding range is the intensity range over which companding action occurs. The companding range of most compandors is usually between 50 db and 60 db. A companding range between these two db limits is usually large enough to provide proper signal-to-noise improvement over the wide intensity range of speech signals. Any high or low speech signal that may appear outside the 50 db to 60 db range is limited without affecting communication messages too much. Notice in figure 19 that companding action takes place over a companding range of 60 db.

HOW, ATTACK AND RECOVERY TIMES

You've learned that the gain or loss imparted to speech signals in a volume compandor is controlled by syllabic variations of the input signal and not by individual speech peaks. Now to make sure that syllabic variations control the gain, time constants are designed into the control circuits of both the compressor and expandor. These time constants are called attack and recovery times.

The attack time is the time it takes to change the power of a speech signal from a low power value to a higher power value. The recovery time is the time it takes to change the power of a speech signal from a high value to a lower value. Normal values for these times are about 3 milliseconds for the attack time and about 13.5 milliseconds for the recovery time.

The attack and recovery times must be properly selected to avoid distorting the speech signals. For example, if the attack time is too slow (less than 3 milliseconds), parts of the speech syllables will be distorted. If the recovery time is too slow, the expandor will not give the proper expansion between syllables. In addition to selecting the proper time constants, it is necessary to synchronize the attack and recovery times of the compressor and expandor. That is, the action of the expandor must follow the reverse action of the compressor, or the speech signals will also be distorted.

FINALLY, WHERE ARE COMPANDORS USED

You'll find compandors used in many types of equipment to reduce the interfering effects of noise and crosstalk. Some specific types of equipments that utilize compandors are telephone carrier equipment, radio equipment, and pulse-code modulation (PCM) multiplexing equipment.

FINAL SUMMARY

In this sheet, you first studied the principles of sound energy related to compandor operation, then you learned what a compandor is, what it does, and how a typical volume compandor works. Now test yourself on how much you have learned by answering the following review items.

Correct answers to these items are given on pages 20 and 21.

REVIEW ITEMS

1. What is the intensity range for a speaker whose lowest power level of speech is 20 db and highest power level is 70 db?

ANSWER: ______________________________

2. What other factor, besides words, determines intensity range?

ANSWER: ______________________________

3. Briefly explain how words help to determine intensity range?

ANSWER: ______________________________

____

4. Briefly explain what is meant by crosstalk.

ANSWER: ______________________________

____

5. List the chief sources of electrical noise.

ANSWER: ____

____

6. Briefly explain how noise affects communications.

ANSWER: ____

7. Briefly explain what effect noise has on strong speech signals.

ANSWER: ____

8. Explain the meaning of a 40 db signal-to-noise ratio.

ANSWER: ____

9. To decrease the ill effects of noise and crosstalk, the signal-to-noise ratio for weak signals must be (increased) (decreased). (Select one.)

10. What two units make up a compandor?

ANSWER: ____

11. What is the purpose of a compandor?

ANSWER: ____

____

12. Physically, where is the compressor unit usually located in a communication system?

ANSWER: ____

13. List the two functions of a compressor unit.

ANSWER: ____

____

14. How does a compressor unit reduce the intensity range?

ANSWER: ____

15. What is the main function of an expandor unit?

ANSWER: ____

____

16. Briefly explain the difference between instantaneous and syllabic compandors.

ANSWER: ____

____

17. List the component parts of a typical volume compandor and give the purpose of each part.

ANSWER:____

18. What is meant by a compression-expansion ratio?

ANSWER:____

19. Explain the meaning of a compression ratio of 2 to 1.

ANSWER:____

20. Briefly explain the meaning of a 50 db companding range.

ANSWER:____

21. Briefly explain what is meant by attack time?

ANSWER:____

22. List the types of equipment in which compandors are used.

ANSWER:____

ANSWERS TO REVIEW EXERCISE ON PAGE 9

1. Frequency and Intensity.

2. 200 Hz to 3,200 Hz.

3. Volume, loudness, and power.

4. Speech intensity range.

5. The stress placed upon the syllable by the speaker.

6. Low power signals become lost in electrical noise. High power signals are a source of crosstalk.

7. Noise.

8. The drowning out of weak signals by noise.

9. Signal-to-noise ratio.

ANSWERS TO FINAL REVIEW ITEMS STARTING ON PAGE 17

1. 50 db.

2. The speaker.

3. Different amount of power is produced by syllables of words when spoken. In turn this difference in power determines the intensity range.

4. Crosstalk is the result of interfering signals "leaking" from one communication channel to another.

5. Lightning, power transmission lines, automotive ignition systems, resistors, transistors, and tubes.

6. Noise drowns out weak speech signals.

7. Noise has little or no effect on strong signals.

8. The signal is 10,000 times as strong as the noise.

9. Increased.

10. Compressor and expandor.

11. A compandor increases the quality of voice transmission by decreasing the effects of interference from noise and crosstalk.

12. In the transmitter section.

13. A compressor raises the power (increases signal-to-noise ratio) of weak signals, and attenuates strong signals.

14. By automatically compressing all speech signals before transmitting the signals.

15. The main function of an expandor unit is to restore the compressed intensity range back to its original range.

16. Instantaneous compandors change the power in speech signals by changing each instantaneous peak value of a speech signal. A syllabic compandor changes the power in speech signals by changing syllabic variations.

17. Variable Loss Device -- determines the overall gain of each syllabic variation of the input signal.

Amplifier -- increases the power of all signals that leave the unit.

Rectifier Control Circuit -- controls the amount of signal attenuation that takes place in the variable loss device.

18. A compression-expansion ratio is a figure which represents the degree to which speech signals are compressed and expanded.

19. A 2 to 1 compression ratio indicates that the speech energy traveling between the compressor and expandor has an intensity range equal to one-half its original value.

20. A 50 db companding range is the intensity range over which companding, or compressing action, takes place.

21. Attack time is the interval of time it takes to change the power of a speech signal from a low power value to a higher power value.

22. Carrier, radio, and pulse-code multiplexing equipment.

APPENDIX A

THE TELEPHONE
TRANSMISSION SYSTEM
SSTS 53002A

OBJECTIVES

1. To explain the makeup of transmission systems.

2. To show you how a telephone transmission system works.

3. To explain the meaning of voice power.

4. To tell you how resistance, leakage, inductance, and capacitance affect the transmission of voice power.

This information sheet supersedes SSTS 53002, The Telephone Transmission System.

TABLE OF CONTENTS

1. INTRODUCTORY INFORMATION

a. The purpose of a telephone system is to transmit voice frequencies ranging from 200 to 4,000 Hertz (Hz). Doing this becomes more and more difficult as we increase the length of the transmission line. The longer we make the line, the less sound we receive at the far end. The line actually limits the distance over which we can transmit.

b. To understand why this happens you have to know more about a telephone transmission system. That's the purpose of this information sheet: to tell you what a transmission system is and to show you how it works.

2. THE PURPOSE OF TRANSMISSION SYSTEMS

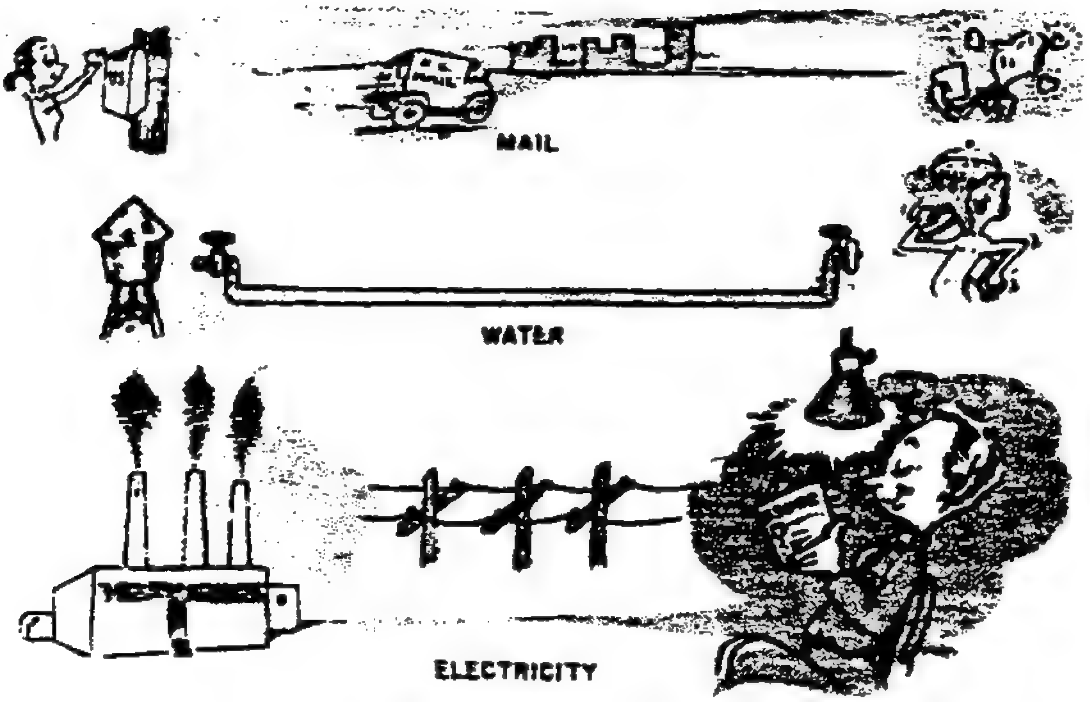
a. Every day you come across many different transmission systems. Here are some examples that you're familiar with (fig. 1): electric, water, mail delivery, bus transportation, trucking and the telephone.

b. All of these systems have one thing in common. They transmit something from one place to another. Of course, each one transmits something different. Still, each system must have the same three main parts to be able to do its job.

3. PARTS OF A TELEPHONE TRANSMISSION SYSTEM

a. Any transmission system has to have a source of energy, a transmission medium, and a receiving device. These are rather high sounding terms for some familiar things. Figure 2 on page 4 gives you an idea of what these things are and why they are needed.

b. The source of energy is needed because it provides what we transmit. The transmission medium is needed to carry what we transmit. Finally, the receiving device is needed to receive what we transmit and change it into some useful form. In the first example shown, the "receivers" convert what they receive into action. In the second, the telephone receiver changes what it receives into sound. Now, let's look at the parts of a telephone system and see how they work.



53002-1

Figure 1. Different transmission systems.

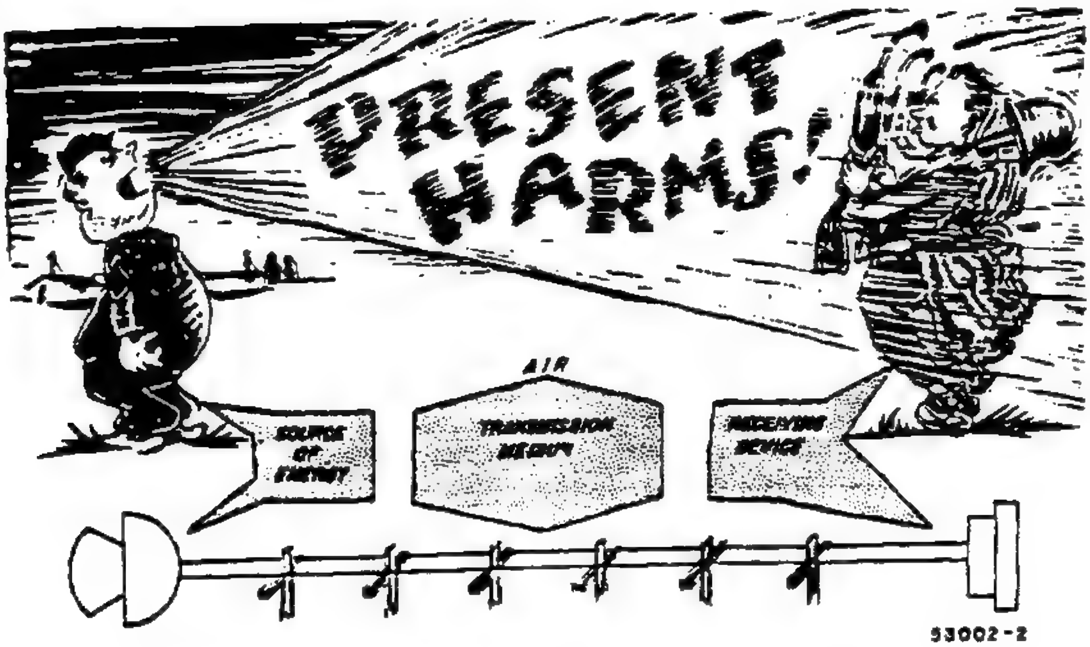


Figure 2. A transmission system needs three parts.

4. SOURCE OF ENERGY IN A TELEPHONE SYSTEM

In a telephone system, the source of energy is the transmitter circuit. It consists of a transmitter, a battery, and an induction coil. This circuit changes sound energy into electrical energy and transfers the electrical energy to the line. In an earlier information sheet you learned how this circuit works. Here we briefly review the circuit operation.

5. SOUND ENERGY IS CONVERTED TO ELECTRICAL ENERGY

a. The transmitter circuit is connected as shown in figure 3. The transmitter acts as an adjustable resistor and controls the current flow from the battery. When no sound is striking the transmitter, steady direct current (dc) flows in the circuit.

b. When sound waves strike the transmitter, the diaphragm bends in and out at the frequency of the sound wave. This in-out movement of the diaphragm makes the resistance of the carbon granules decrease and increase at the same frequency as the sound wave. A flow of fluctuating dc results. And, since the fluctuating dc flows through the primary of the induction coil, alternating current (ac) voltage is induced across the secondary.

c. The secondary of the coil is connected to the line. And the induced ac voltage is, therefore, applied directly across the line. The induced ac voltage causes a flow of alternating current in the line and through the receiver at the other end. This alternating current is the same frequency as the original sound wave that started the action.

d. The ac voltage across the line and the current flowing in the line make up the electrical energy that we call voice power.

6. STANDARD AMOUNT OF TRANSMITTED VOICE POWER

a. When a person talks into a telephone in a clear, loud voice, a very small amount of electrical energy is produced. This energy is in the form of ac voltage and current. Both of these (voltage and current) are so small that we usually don't speak of them by themselves. Instead, we speak of the combination of the voltage and current, which is power.

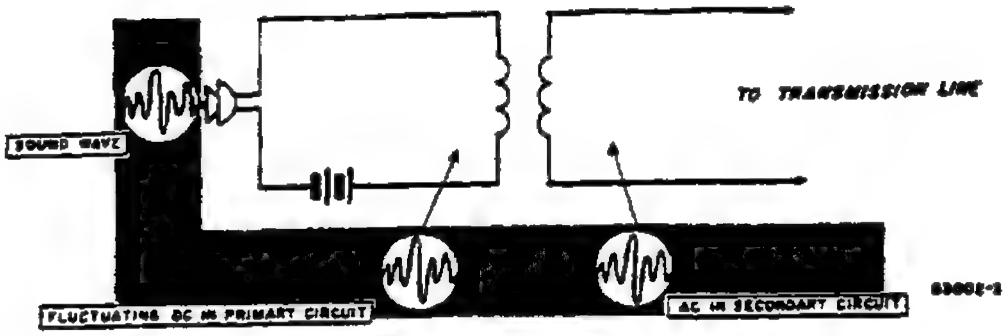


Figure 3. Telephone transmitter circuit.

b. You remember from your study of Ohm's Law that power is the product of voltage and current. In other words, power in watts can be found by multiplying voltage by current (P = E x I). Working this formula out for the power produced by a voice sound in a telephone gives us something like this:

(1) The voltage across the induction coil secondary is about 1 volt.

(2) The current caused by this voltage is about .001 ampere (1 milliampere).

(3) And since power in watts is:

P = Voltage (E) x Current (I)

P = 1 volt x .001 ampere

P = .001 watt (1 milliwatt)

c. The voltage and current values given in the formula are approximate. These values vary according to the loudness of the voice. But the value of 1 milliwatt is considered as the standard for voice power produced in a telephone. And you'll be dealing with this standard milliwatt from now on.

7. VOICE POWER IS CONVERTED INTO ELECTRICAL POWER

As we said before, your job is to transmit sound. But you hardly ever deal with sound in its natural form. When you get the sound, it's usually in the converted form of electrical power. And this power -- this 1 milliwatt -- is extremely small. It's so small that it would take 60,000 milliwatts to light one 60-watt lamp bulb. Yet, despite its smallness, this milliwatt of voice power has to go from one end of a transmission line to the other. And you have to see that it gets there.

8. TRANSMISSION MEDIUM AFFECTS VOICE POWER

a. The transmission medium of a telephone system is the part that carries voice power from the transmitter to the receiver. The transmission medium is the telephone line.

b. A telephone line (fig. 4 on page 6) consists of two wires separated by some type of insulation. This is true whether the wires are in a cable or out in the open. Bare open wires, like those you see strung on telephone poles, are insulated from each other by air, while wires in a cable are insulated with a paper, rubber, cotton or other type of covering.

c. Regardless of type, however, all lines perform the same job. They all affect the voice power in about the same way. By this we mean that all telephone lines reduce the amount of voice power that we are trying to transmit. In other words, you can never expect to receive the same amount of voice power at the receiver as you start out with at the transmitter. The telephone line always reduces the voice power that it is carrying. This reduction of voice power is called attenuation.

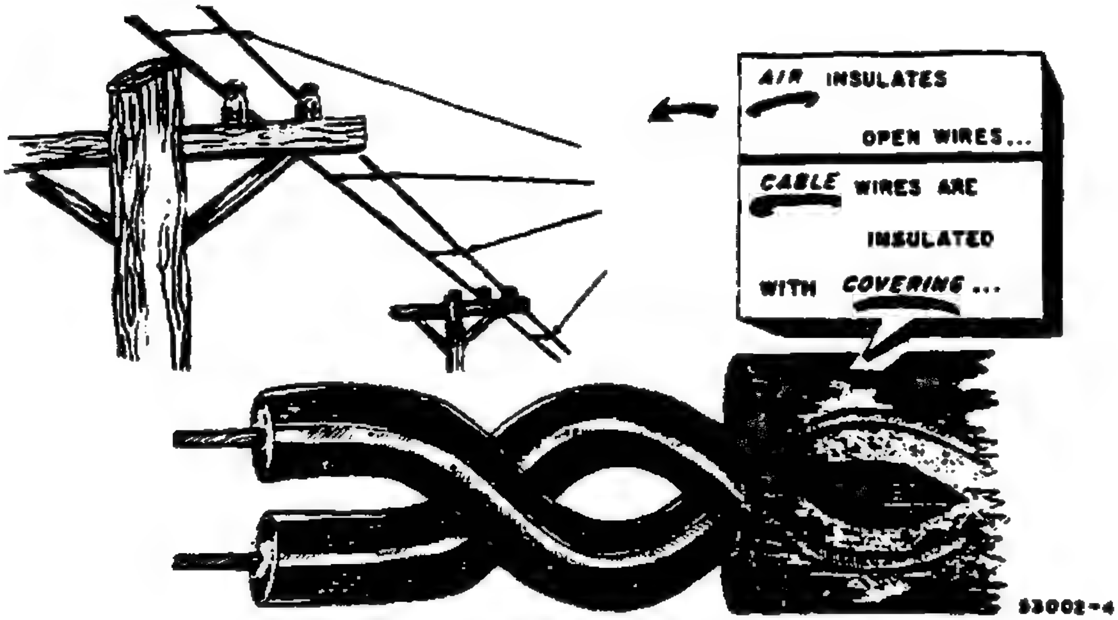


Figure 4. Types of telephone wires.

9. DEFINITION OF ATTENUATION

a. Attenuation comes from the verb attenuate which means to reduce, cut down, lessen, thin out, decrease, or weaken. Although we don't hear the word used in everyday language, it could be used in talking about everyday things. For instance, it would be correct to say, "If you do not read this text carefully, your grade for this lesson may be greatly attenuated."

b. In your work, you'll see and hear these words used very often:

(1) ATTENUATE -- which means REDUCE or CUT DOWN

(2) ATTENUATION -- which means REDUCTION

(3) ATTENUATOR -- which is a type of circuit that REDUCES or CUTS DOWN.

c. Know what these words mean when you come across them. Begin to use them yourself. They are as much a part of your work as the words volts and ohms.

10. TELEPHONE LINES ATTENUATE VOICE POWER

a. Making this statement is something like asking, "Why does a road cause automobile tires to wear out?" Everyone knows that it's the constant friction of the tires against the road surface that wears them out. It's just about the same with the telephone line. The constant friction of the wires wears away the voice power.

b. There are two different kinds of friction in these examples. On the road it's mechanical friction. In the telephone line, it's electrical friction caused by Resistance (R), Leakage (G), Inductance (L), and Capacitance (C) (fig. 5).

c. These four are present in all telephone lines. Because of this, they are called the four properties or the four characteristics of a transmission line. Each one has a different effect on voice power. We'll consider each one separately in the paragraphs that follow.

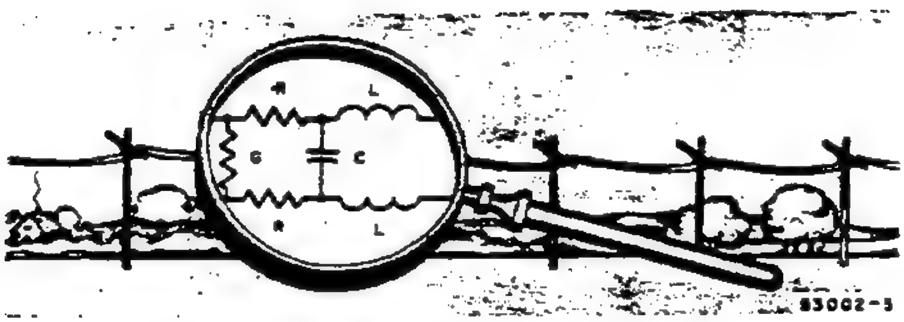


Figure 5. Looking at a telephone line more closely.

11. RESISTANCE (R) OF CONDUCTORS

a. All conductors have resistance. A piece of wire no larger than a pencil lead has a definite amount of ohmic resistance. Even though this piece of wire is extremely small, you can still measure the resistance.

b. Here's an example. One type of wire used in telephone lines has a resistance of about .00066 ohms per inch (fig. 6). This is a very small value and you might think we could forget about it. But suppose you put about 63,330 inches of this wire together into one length. It would equal one mile of wire. The resistance for this mile would be: .00066 ohms times 63,330 inches, which is about 43 ohms.

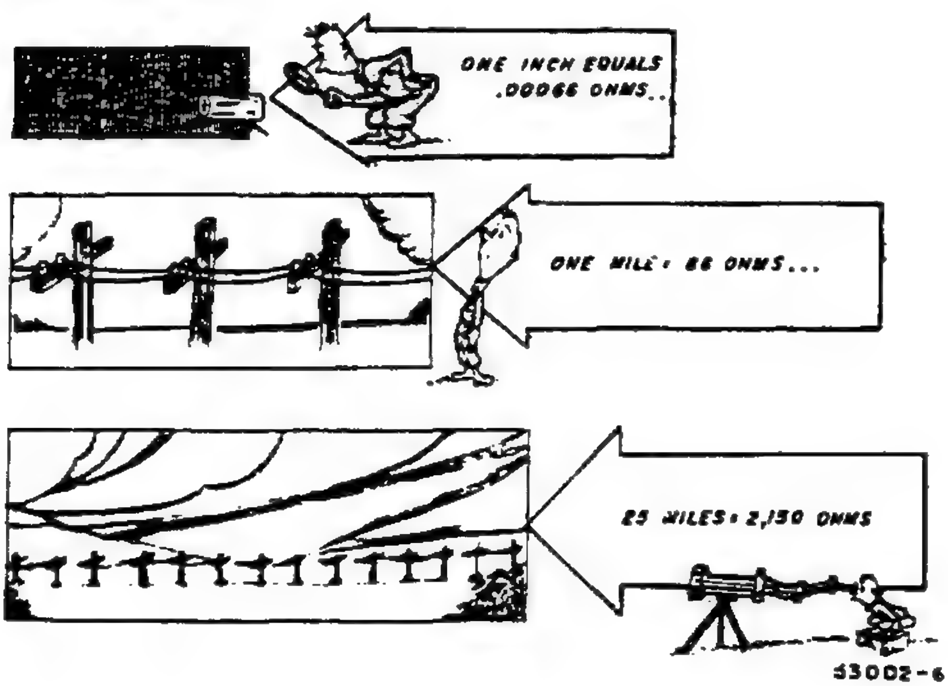


Figure 6. Telephone lines have resistance.

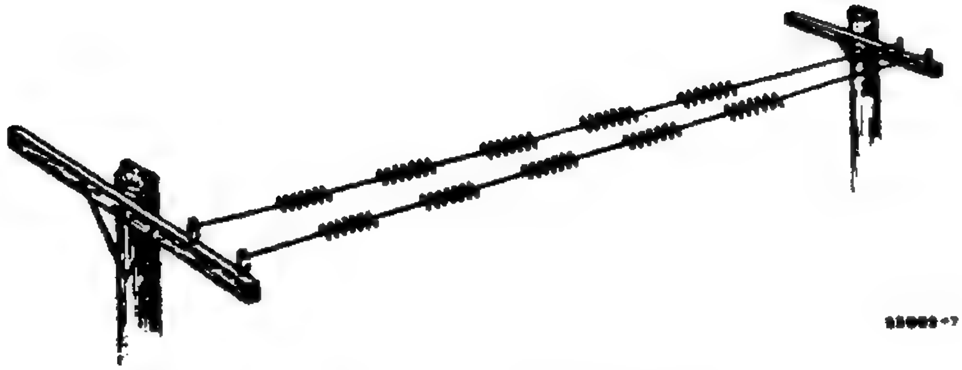


Figure 7. Line wires act as a series-resistance circuit.

12. USING WIRE WITH RESISTANCE IN A TELEPHONE LINE

a. Now suppose you wanted to use the same type of wire to construct a one-mile telephone line. You'd have to get two pieces one mile long, because a telephone line needs two wires. The resistance of each wire would be 43 ohms. This means that the total resistance of the one-mile line would be twice 43 ohms or 86 ohms. Remember, this 86 ohms is the resistance for only one mile of line. Lines, however, are much longer than this. If this same type of wire is used in a 25-mile circuit (which is more like it) the resistance would be 2,150 ohms (86 ohms x 25 miles).

b. Simply stated, the two wires of a telephone line (fig. 7) act as two very long resistors whose resistance increases as the length of the line increases. The transmitter circuit is connected to one end of these "resistors," and the receiver circuit is connected to the other end. These "resistors" attenuate the voice power. But before we explain exactly how this attenuation takes place, we want to talk about another type of resistance that is present in a telephone line. This is the resistance between the two wires which causes leakage (G).

13. LEAKAGE (G) THROUGH INSULATORS

a. There are no perfect insulators. All materials, including rubber, paper, wood, glass, and air are really conductors. Of course, they are very poor conductors; that's why they are called insulators. And, for many practical purposes, these materials act as perfect insulators -- but not in a telephone line.

b. In a line, insulating materials are used to prevent current from flowing between the wires (from one wire to the other). But because these materials are not perfect, they can't stop all the current flow. And, therefore, the insulators themselves provide a high resistance path for current flow between the wires. This high resistance is called insulation resistance. And the current flowing through the insulation is called leakage. The symbol for leakage is the capital letter G. Leakage occurs in all types of telephone lines, whether they are bare open wires or insulated cable wires.

14. LEAKAGE IN OPEN WIRE LINES

a. The insulation used between the wires of open wire lines is air -- a fairly good insulator. Nevertheless, it provides a path for current flow. The air between the wires acts as a lot of resistors connected in parallel between the wires along the full length of the line (fig. 8).

b. In the figure, you also see resistors connected from the two wires to ground. These resistors represent the path through which small amounts of current leak to ground. Leakage to ground happens in different ways:

(1) Current leaking directly through the air down to the earth.

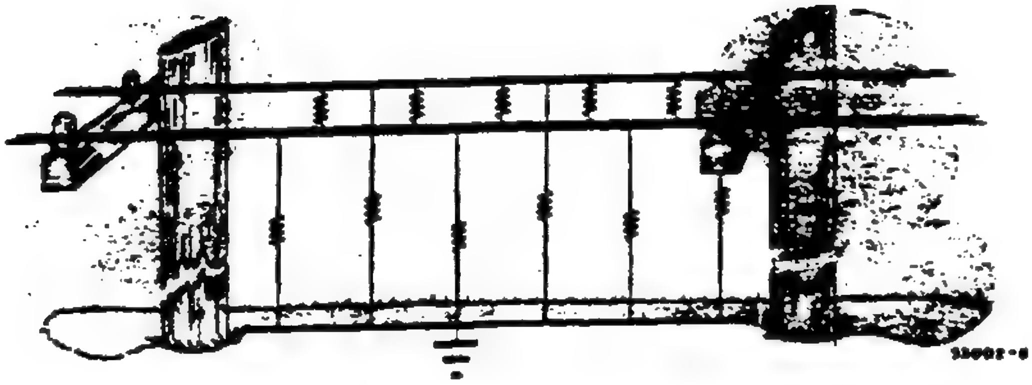


Figure 8. Air acts as parallel resistors.

(2) Leakage at points where the wires are attached to cross-arms of telephone poles. Current leaks from the wire, down through the pole and into the ground.

(3) Leakage of current through tree branches (which may touch or be near the wires) down through the tree trunk to ground.

c. An important point to remember about leakage in open wire lines is that weather conditions affect it greatly. In damp or wet weather, the air between the wires becomes a much poorer insulator. The insulation resistance goes down and there is more leakage. This doesn't happen in cables because the wires are sealed in where moisture can't affect them. But we still get leakage in cables.

15. LEAKAGE IN CABLES

a. In cables, the wires are close together. The insulation around each wire touches the insulation of another wire. This insulation has resistance (fig. 9 on page 10) just like the air in open wire lines. And this resistance is also very high -- usually higher than the resistance of air. Therefore, we still get leakage between the wires. And we still get some leakage to ground.

b. Leakage to ground happens in cables because there is usually a metal covering around the cable, which is connected to ground. In some cables, this covering is on the outside and it's made of lead. In others, the metal covering is in the form of a metal braid (a basket weave) which is under an outside rubber or plastic covering. This last one, by the way, is the type of cable that you'll use with carrier equipment (spiral-four cable).

16. PRACTICAL IDEAS ABOUT LEAKAGE

a. In discussing leakage, we've said that the insulation resistance which causes leakage is very high. This phrase "very high" doesn't really say much. So to give you an idea of how high insulation resistance is, here are a few facts: Insulation resistance is so high that it is measured in megohms (millions of ohms), and it decreases as the length of the line increases.

b. For example, the insulation resistance of the spiral-four cable we mentioned earlier is 200 megohms (200 million ohms) for one mile. But the insulation resistance goes way down to 8 megohms when we increase the length of the cable to 25 miles. Each mile is like a 200-megohm resistor connected in parallel. By using Ohm's Law we find that twenty-five 200-megohm resistors connected in parallel is equal to 8 megohms.

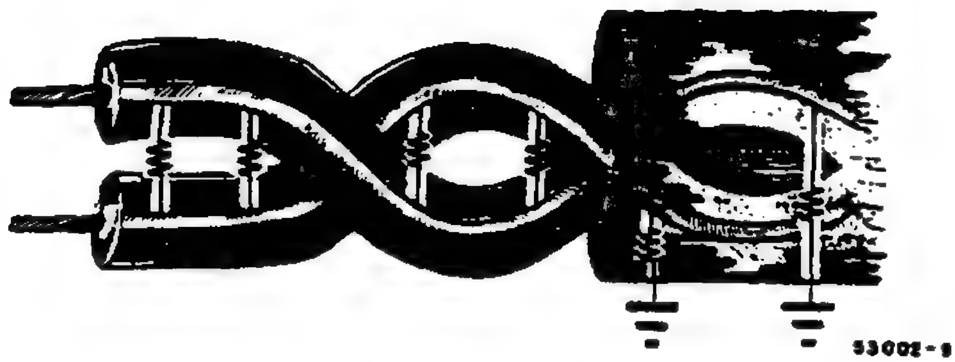


Figure 9. Leakage in cables.

c. You can see that leakage will increase as the line is made longer. It's as simple as this: Longer line = lower insulation resistance = more leakage.

d. Now here's the important thing about leakage. The current leaking between the wires and to ground is part of voice power. It's part of the 1 milliwatt that you're trying to send to the receiver. And this loss caused by leakage, plus the loss caused by resistance of the wires, is attenuation.

17. RESISTANCE AND LEAKAGE CAUSE ATTENUATION

a. Now you know what is meant by the resistance and leakage properties of a line. The next question is: How do these properties attenuate the voice power?

b. Assume that you have sent out from a transmitter 1 mw (1 milliwatt) of voice power. Assume further that this 1 mw is made up of 1 volt and 1 milliampere (ma.) (1 volt x .001 ampere = .001 watt). At the far end of the line you would like to receive the same amount of voltage, current and power as shown in figure 10. You know, however, that this is impossible because of the resistance of the wires and insulation resistance. Instead of getting the same amount of power out, you'd get something like what you see in figure 11.

18. RESISTANCE CAUSES A VOLTAGE DROP

The resistance of the wires causes the voltage applied at the transmitting end to drop. Every tiny millionth of an inch of wire along the way causes the voltage to drop a little more. This means that the voltage across the receiver is less than the voltage applied at the transmitter.

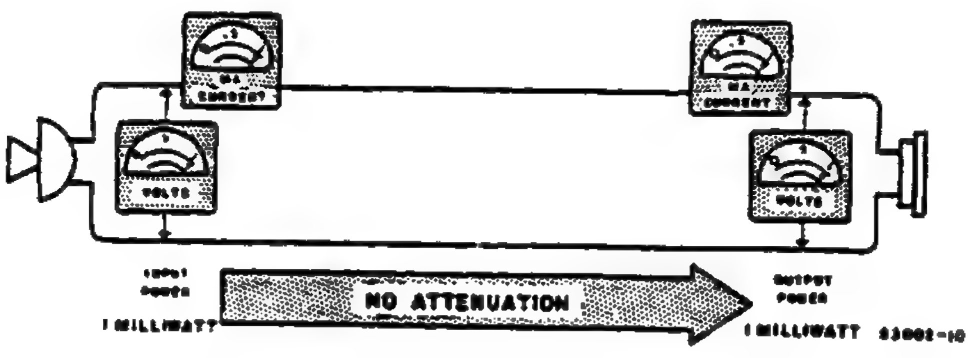


Figure 10. Ideal transmission system.

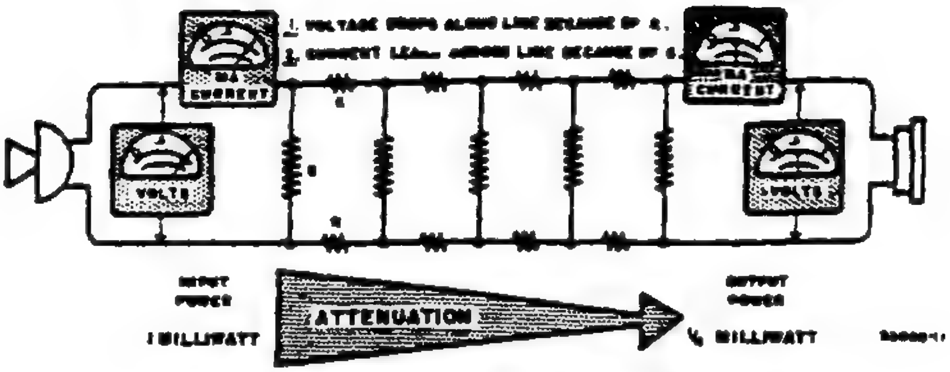


Figure 11. Power received is less than power sent.

19. LEAKAGE RESULTS IN A CURRENT LOSS

Figure 11 also shows that all of the 1 ma. of current supplied from the transmitter does not reach the receiver either. All along the line, current leaks through the insulation and returns to the transmitter. This leakage-current does not reach the receiver. So, flowing through the receiver, there is less current than the 1 ma. that started out from the transmitter.

20. LESS VOLTAGE AND LESS CURRENT MEANS LESS POWER.

a. The current and voltage at the receiving end are less than the transmitted voltage and current. This results in less voice power at the receiver, since P = E x I. You can see this in figure 11 where 1 mw is sent but only 1/4 mw is received. Three-fourths or 75 percent of the voice power is lost in this circuit because of resistance and leakage. That's a lot of attenuation. Still, it's only part of the total attenuation that we get in a line.

b. The other two line properties, inductance and capacitance, also attenuate the voice power.

21. INDUCTANCE (L) AND CAPACITANCE (C) CAUSE ATTENUATION

a. Because of these two properties, a line causes attenuation which is not the same for all frequencies. The attenuation gets greater as the transmitted frequencies go higher. This means that a line reduces the voice power of people who speak at high frequencies more than those who speak at low frequencies. Also, when people change from low to high frequencies while telephoning, they are not heard as well at the high frequencies, even though they talk just as loud both times. This is shown graphically in figure 12 on page 12.

b. The graph on the left shows that all frequencies from 200 to 4,000 Hertz are transmitted at the same power level of 1 mw. The graph on the right shows that all frequencies are received at different power levels. You'll notice that less power is received for each higher frequency. This is called unequal attenuation, and results in distortion. In other words, the line does not give faithful reproduction because of inductance and capacitance. Why do these two properties cause unequal attenuation? Let's study them one at a time and find out.

22. INDUCTANCE (L) OF TELEPHONE LINES.

a. A line has inductance because of the ac voice currents flowing through the wires. Here's what happens:

(1) Ac voltage from the transmitter causes ac voice current flow in the line

(2) The ac voice current flow produces a changing magnetic field around the wires.

(3) The changing magnetic field induces voltages into the wires all along the line.

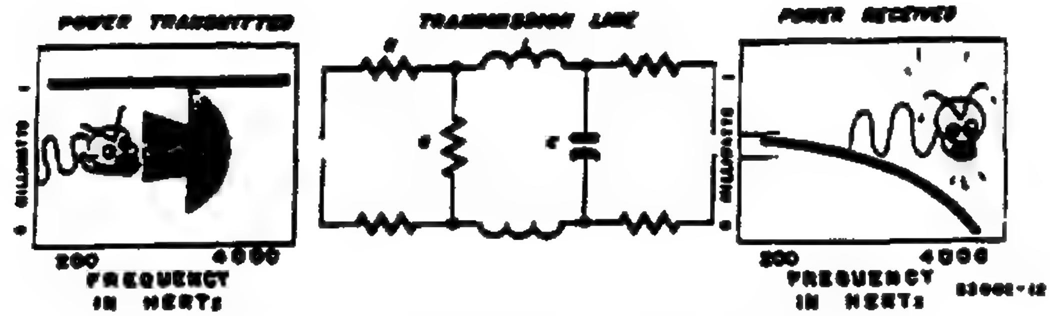


Figure 12. Inductance and capacitance cause unequal attenuation.

(4) The induced voltages oppose the applied voltage that is causing the current flow.

(5) This results in gradual reduction of voltage along the line, as shown in figure 13.

(6) Less voltage gives less current all along the line too, since current is a direct result of voltage.

b. What this all adds up to is a loss of voice power. Less voltage and less current reach the receiver and, therefore, less power is delivered to the receiver.

23. VOLTAGE, CURRENT, AND POWER LOSSES INCREASE WITH FREQUENCY

This is easy to see if you remember this: The opposing voltages induced into the wires (as a result of voice current flow) become greater as the magnetic field speeds up its movement. And since this happens when voice current flows at a higher frequency, there is naturally more opposition to the applied voltage at this time. So, for each higher frequency transmitted, there is less voltage, less current, and less power all along the line.

24. INDUCTIVE REACTANCE IS DETERMINED BY A FORMULA

a. The opposition caused by inductance is called inductive reactance (X_L). This is expressed in ohms just like dc resistance. To find out how much inductive reactance there is in a circuit, we use the formula $X_L = 2\pi FL$. You've seen and used this formula before. And you know that as F (frequency) is increased, X_L also increases as long as L (inductance) is not changed.

b. The formula has been worked out for nine different frequencies in Table I. There it's plain to see that with an inductance of .006 henries, the opposition (X_L) increases from 7.5 ohms at 200 Hertz all the way up to 150.7 ohms at 4,000 Hertz. Now, since you know that greater opposition means greater loss, you can see why inductance causes unequal attenuation.

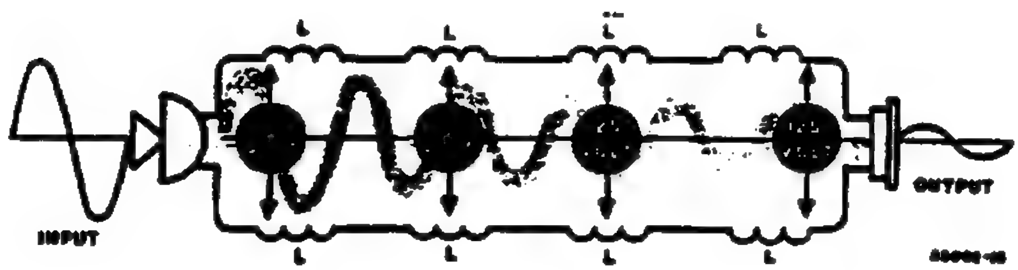


Figure 13. Inductance reduces voltage.

TABLE I

X_L (Inductive Reactance)		2π	x	F (Frequency)	x	L (Inductance)
7.5 ohms	=	6.28	x	200 Hertz	x	.006 henries
11.3 ohms	=	6.28	x	300 Hertz	x	.006 henries
15.1 ohms	=	6.28	x	400 Hertz	x	.006 henries
37.6 ohms	=	6.28	x	1,000 Hertz	x	.006 henries
75.4 ohms	=	6.28	x	2,000 Hertz	x	.006 henries
113.0 ohms	=	6.28	x	3,000 Hertz	x	.006 henries
120.6 ohms	=	6.28	x	3,200 Hertz	x	.006 henries
131.8 ohms	=	6.28	x	3,500 Hertz	x	.006 henries
150.7 ohms	=	6.28	x	4,000 Hertz	x	.006 henries

25. TELEPHONE LINES HAVE CAPACITANCE (C)

a. A capacitor consists of two conductors separated by some type of insulation. This is what you have in a telephone line, two long conductors separated by insulation. This holds true for open wire as well as cable lines.

b. The capacitance in a line (fig. 14), as you can see, is present between the wires for the full length. It's just as if there were small capacitors connected between the wires at every point along the line. This is what we mean by the capacitance (C) property of a line.

c. You already know that capacitance (C) causes unequal attenuation of voice power. Now we want to find out how this happens.

26. HOW CAPACITANCE (C) AFFECTS VOICE POWER

a. Capacitance provides a path for current to flow between the wires of the line. This path offers less opposition as the frequency increases.

b. In other words, as the frequency of the voice currents being transmitted increases, more and more current is shunted from one wire to the other. Of course this doesn't mean that current flows through the capacitance path. It means that since voice currents are ac, they effectively flow through the capacitance path just as ac effectively flows through any capacitor.

c. Now since this shunting effect of capacitance becomes greater with a higher frequency, less current is delivered to the receiver at the higher frequency. We can say, then, that capacitance causes attenuation which increases as the frequency rises.

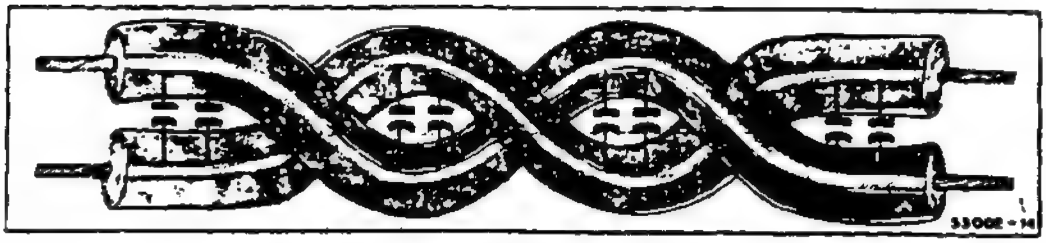


Figure 14. A telephone line has capacitance.

TABLE II

X_C (Capacitive Reactance)	=	$\dfrac{1}{2\pi \times F\ (\text{Frequency}) \times C\ (\text{Capacitance})}$
796 ohms	=	$\dfrac{1}{6.28 \times 200\ \text{Hertz} \times .000001\ \text{farad}}$
531 ohms	=	$\dfrac{1}{6.28 \times 300\ \text{Hertz} \times .000001\ \text{farad}}$
398 ohms	=	$\dfrac{1}{6.28 \times 400\ \text{Hertz} \times .000001\ \text{farad}}$
159 ohms	=	$\dfrac{1}{6.28 \times 1,000\ \text{Hertz} \times .000001\ \text{farad}}$
79 ohms	=	$\dfrac{1}{6.28 \times 2,000\ \text{Hertz} \times .000001\ \text{farad}}$
53 ohms	=	$\dfrac{1}{6.28 \times 3,000\ \text{Hertz} \times .000001\ \text{farad}}$
50 ohms	=	$\dfrac{1}{6.28 \times 3,200\ \text{Hertz} \times .000001\ \text{farad}}$
46 ohms	=	$\dfrac{1}{6.28 \times 3,500\ \text{Hertz} \times .000001\ \text{farad}}$
40 ohms	=	$\dfrac{1}{6.28 \times 4,000\ \text{Hertz} \times .000001\ \text{farad}}$

27. CAPACITIVE REACTANCE IS DETERMINED BY A FORMULA.

a. The capacitance path for current flow between the wires of a line decreases its opposition as the frequency rises. This opposition is called capacitive reactance (X_C). And the amount of opposition (X_C) to currents of any frequency can be found by the formula:

$$X_C = \frac{1}{2\pi FC}.$$

b. You have seen and used this formula. And you know, that if C (capacitance) is held constant while F (frequency) is increased, X_C will decrease. Examples of this are given in Table II. In the table the capacitance is held at 1 mf (.000001 farad) while the frequency is varied from 200 to 4,000 Hertz.

c. From the table you can see that capacitive reactance (X_C) reduces its opposition from 796 ohms at 200 Hertz to only 40 ohms at 4,000 Hertz. This means, of course, that more current is effectively shunted across the line at 4,000 Hertz than at 200 Hertz, and, therefore, less current reaches the receiver at 4,000 Hertz. Less current naturally means less power. You can see, therefore, why capacitance causes attenuation which increases as the frequency rises.

28. THE FOUR ELECTRICAL PROPERTIES OF TELEPHONE LINES

a. You realize now that the transmission medium for a telephone system is not as "innocent" as it looks. The simple wires that do the job have more in them than meets the eye. And these hidden properties cause attenuation and what's even worse, unequal attenuation. Let's briefly review these important line properties again.

(1) First of all, remember that you're transmitting 1 milliwatt of voice power. This is applied at the transmitter in the form of small amounts of voltage and current.

(2) The telephone line which carries this 1 milliwatt is really a long circuit (fig. 15) composed of the following:

(a) Series RESISTANCE (R).

(b) Parallel (insulation) resistance which causes LEAKAGE (G).

(c) Series INDUCTANCE (L), and

(d) CAPACITANCE (C) between the wires.

(3) Each one of these properties reduces the voice power.

(4) RESISTANCE causes the applied voltage to drop all along the line, resulting in less voltage, and less power at the receiver.

(5) LEAKAGE is the current that flows from one wire (through the insulation) to the other wire. This happens all along the line too. The current that leaks is not delivered to the receiver. There is, therefore, less current and less power at the receiver.

(6) INDUCTANCE causes opposition (X_L) that reduces the applied voltage all along the line. This opposition increases as the frequency rises ($X_L = 2\pi FL$). This gives increasingly less voltage, less current and less power at the receiver as the frequency rises.

(7) CAPACITANCE effectively shunts current from one wire to the other all along the line. More current is shunted at higher frequencies because the opposition to current flow (X_C) decreases as the frequency rises ($X_C = \frac{1}{2\pi FC}$). This causes increasingly less current and less power at the receiver as the frequency rises.

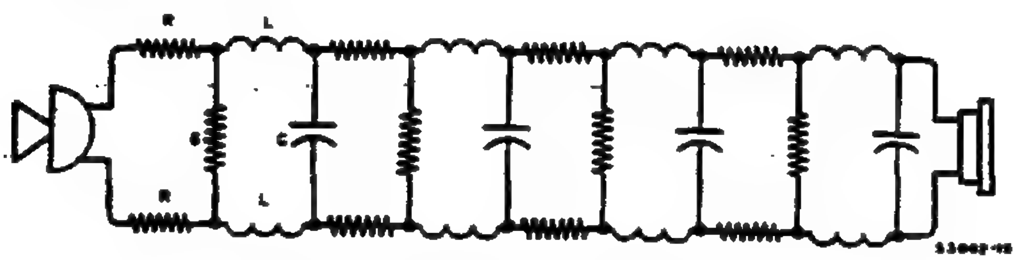


Figure 15. Telephone line properties.

b. Finally, then, we say that the line properties (R, G, L, C) attenuate the voice power. And the attenuation increases as the frequency rises. In other words, the line reduces the loudness of all sounds transmitted and this reduction increases as sounds get higher in frequency.

29. ONE MILLIWATT OF POWER IS NEEDED TO TRANSMIT SOUND

All that's been said about attenuation probably has you wondering how you ever hear anything over the telephone. Well, the situation isn't as bad as it may seem. Even though we start out with only 1 milliwatt, we can lose quite a bit and still get sound from the receiver.

30. MAXIMUM ALLOWABLE POWER LOSS IN A CIRCUIT

a. The receiver, as you learned earlier in this information sheet, converts electrical power back to sound. It does this by electromagnetic action. The ac voice currents flowing through the receiver windings cause the diaphragm to move in and out at the same frequency. This, in turn, sets up air vibrations producing the sound originally applied at the transmitter.

b. Now the receiver can't produce any sound unless it gets a certain amount of power. This means that there is a limit to how much power can be lost in a circuit and still get sound out of the receiver. The limit for military telephone circuits is a loss of 99.9 percent of the .001 watt (1 mw) sent from the transmitter (fig. 16).

c. This may seem surprising, but it's true. A telephone receiver can produce sound if it receives only .000001 watt (1 microwatt) of voice power. Another way of looking at it is that you can lose 999 parts of the original 1 milliwatt sent from the transmitter and still have enough power left.

d. This, of course, is the bare minimum. It isn't a goal you're trying to shoot for. You want to receive much more power than this 1 microwatt, because the more power received, the louder the sound. The limit you do shoot for is a reduction in loss of 75 percent of the original voice power. When 1 milliwatt of voice power is sent out you try to arrange the circuit so that 1/4 milliwatt (25 percent of the original voice power) is received. It has been found that this amount gives satisfactory sound reproduction. Read these values again so you can remember them.

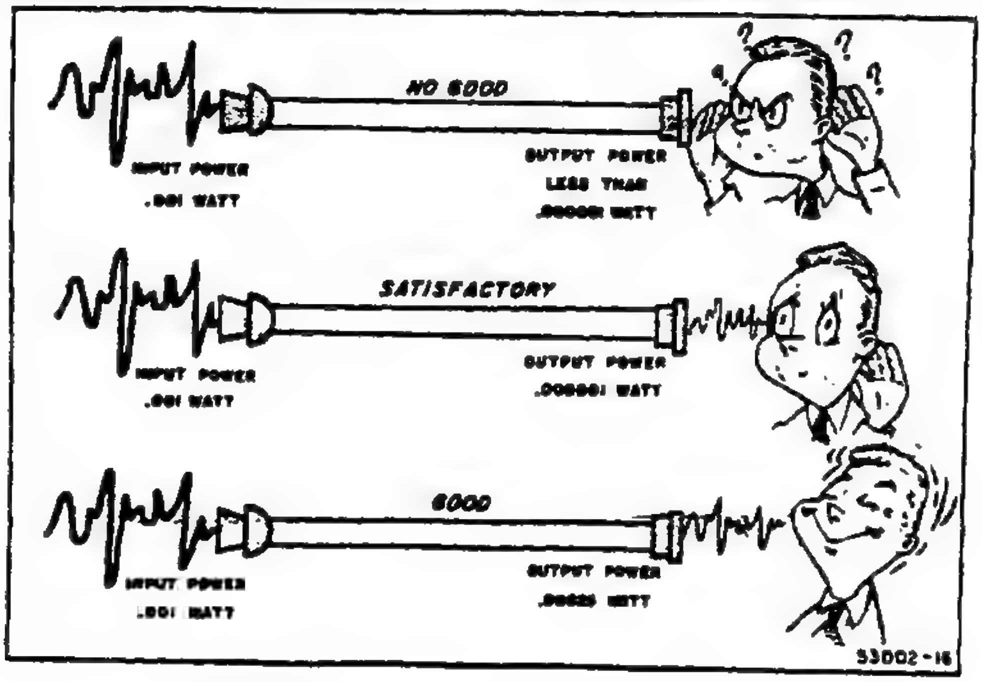


Figure 16. Comparison of voice power received.

(1) Voice power sent from the transmitter is .001 watt (1 milliwatt).

(2) Maximum loss allowable in military circuits is 99.9 percent giving a received power of .000001 watt (1 microwatt).

(3) A reduction in loss of 75 percent of the original voice power is much more desirable since it gives received power of .00025 watt (1/4 milliwatt).

31. SUMMARY

a. You read at the beginning of this sheet that a line limits the distance over which you can transmit. Now you know why this happens. You know that attenuation is what causes this limit. And you realize that attenuation and unequal attenuation are the big problems in long lines transmission.

b. As you go on, you'll learn about other transmission problems. But, to understand them, you must remember what you've learned here. So let's briefly review the main points once more.

(1) A telephone transmission system consists of three main parts:

(a) A source of energy which is the transmitter circuit.

(b) A transmission medium which is the telephone line.

(c) A receiving device which is the receiver circuit.

(2) The transmitter circuit changes sounds of many different frequencies to electrical energy having the same frequencies.

(3) This electrical energy is called voice power and the standard unit for transmitted voice power is .001 watt (1 milliwatt).

(4) Voice power is carried by the telephone line from the transmitter to the receiver.

(5) The telephone line reduces (attenuates) the voice power which it carries.

(6) This attenuation is caused by the line properties, Resistance (R), Leakage (G), Inductance (L), and Capacitance (C).

(7) The line properties cause attenuation which increases as the frequency rises (unequal attenuation). This means that, as the frequency rises, increasingly less voice power reaches the receiver.

(8) The receiver takes the voice power from the line and converts it back to sound.

APPENDIX B

OVERCOMING TRANSMISSION LINE ATTENUATION

IT 53004B

TABLE OF CONTENTS

Note: This information text supersedes SSTS S3004A, Overcoming Transmission Line Attenuation.

1. INTRODUCTORY INFORMATION

a. There are several characteristics of transmission lines that cause a signal to be attenuated. However, the signal in a long transmission line may not be completely lost because we can either overcome or reduce the affects of attenuation. This text covers two methods of combatting attenuation -- using repeaters and using loading coils. The difference between the two methods is shown in figure 1. A repeater overcomes attenuation at the repeater site because it amplifies the signal. A loading coil reduces the effect of the shunt capacitance of the line. By reducing the shunt capacitance, attenuation is also reduced and more signal power can get through.

b. Signal power loss is primarily the result of line (I^2R) loss. To reduce the I^2R loss you can either reduce line resistance (R) or line current (I). To reduce the line resistance would require the use of thicker wire which is both impractical and very expensive. To reduce the line current (I), you would have to increase the impedance of the transmission line.

2. COMBINING INDUCTANCE AND CAPACITANCE IN A LINE

a. Inductive reactance (X_L) and capacitive reactance (Xc) have an opposing and cancelling effect upon each other. When X_L and Xc are equal in value, they completely cancel each other. According to the equation $Z = \sqrt{R^2 + (X_L - X_C)^2}$, when X_L = Xc the impedance becomes resistive (Z = R) and is reduced. This condition is called resonance and is usually desirable in a circuit.

b. A telephone transmission line is one of the circuits where resonance is not desirable because it may cause howling or singing at the resonant frequency points. However, for the purpose of power transmission, it is desirable that X_L cancel out most of the Xc.

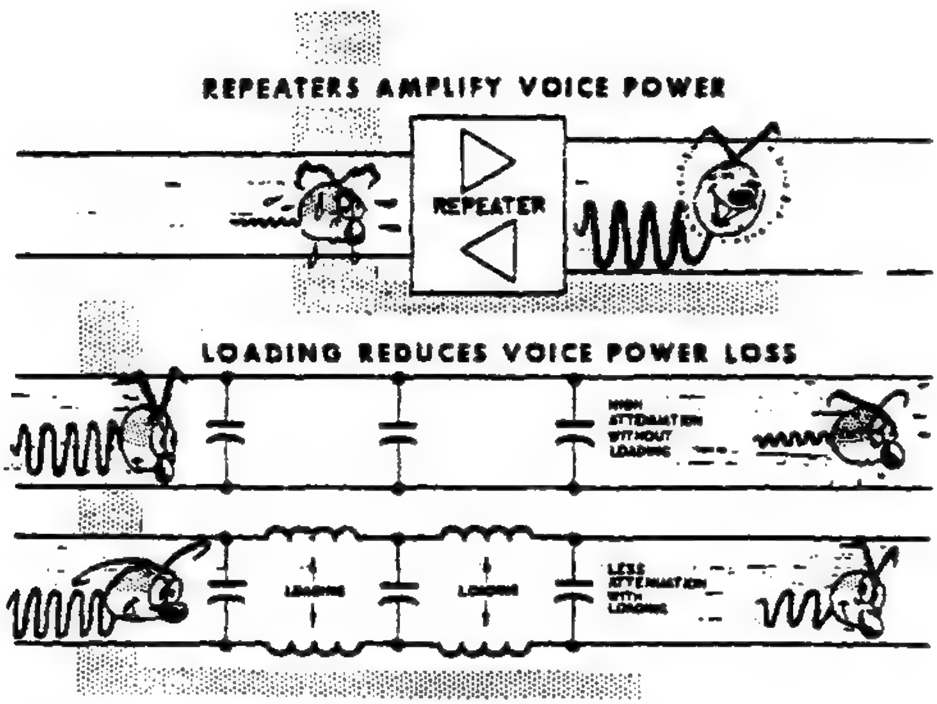


Figure 1. Repeaters and loading work differently.

c. To determine the impedance of a telephone line we can use the formula:

$$Z = \sqrt{\frac{R + j\omega L}{G + j\omega C}}$$

Although the formula may look somewhat complicated, we can use it to show the result of adding inductance in a transmission line. Any inductance (L) that we add will also include a resistance (R) factor. Both of these quantities are in the numerator, under the square root sign. If you increase the numerator of the fraction (R and L) without increasing the denominator (capacitance, C), the resultant impedance, Z, will be increased. This means that when you add inductance to a transmission line, you reduce the effective capacitance and at the same time increase the impedance of the line.

d. The increased impedance automatically reduces the line current and this reduces the I^2R loss of the line. The reduction in line current also reduces the possibility of, or prevents, singing or howling. Because the current is reduced, a higher voltage is required to transmit the signal power. Thus, adding inductance to a transmission line produces a "transformer effect" on the line -- the signal is sent out with high voltage, low current, and has low I^2R loss or attenuation.

3. LOADING MEANS ADDING INDUCTANCE

We say that we load a line when we add inductance to it. To add inductance, loading coils are connected in series with a line as shown in figure 2. Two transmission lines are shown in figure 3. One transmission line is loaded and the other one is not. At the left is a graph of the voice frequency input to the line. Notice that all of the voice frequencies have the same power level at the input to the line. At the right of each line is a graph showing how these voice frequencies have been attenuated by the transmission line. The graphs show that the output of the loaded line has less attenuation than the output of the nonloaded line. This means that the voice frequency signals can be sent over a greater distance on loaded transmission lines.

4. LOADING ALSO REDUCES DISTORTION

a. Look at the graph for the nonloaded line (fig 3) and you can see that attenuation increases with a rise in frequency. The higher frequencies are attenuated more than the lower frequencies which is a form of distortion. The distortion is caused by the effective capacitance of the line. This happens because the capacitive reactance decreases as the frequency increases; more high frequency current is shunted from

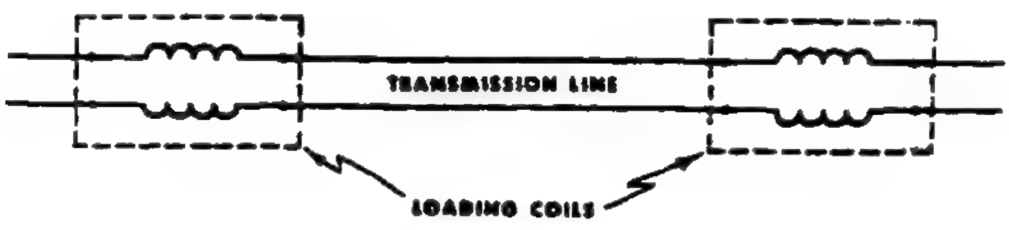


Figure 2. How loading coils are connected in a line.

one wire to the other and less of it gets to the receiver.

b. Loading reduces the capacitive effect. The graph at the end of the loaded line (fig 3) shows that the attenuation is relatively flat over a large portion of the frequency range. Thus, the voice currents transmitted over a loaded line have less distortion and the voices from the telephone receiver have a more natural sound.

5. LOADING A TRANSMISSION LINE

The effective capacitance of a transmission line is distributed along the entire line; it is not lumped at any one point. It's as if there were thousands of small capacitors connected in parallel along the entire length of the line (fig 4). The distributed capacitance has a shunting effect on voice currents all along the line. Since inductive loading is used to combat the capacitive shunting effect, the loading coils must also be distributed along the entire line. If only one large loading coil were connected in the center, at the end, or anywhere else along the line, it would cause a condition called overloading. An overloaded line causes more attenuation and more distortion than a line without any loading at all. The reason is that the large inductance in a single

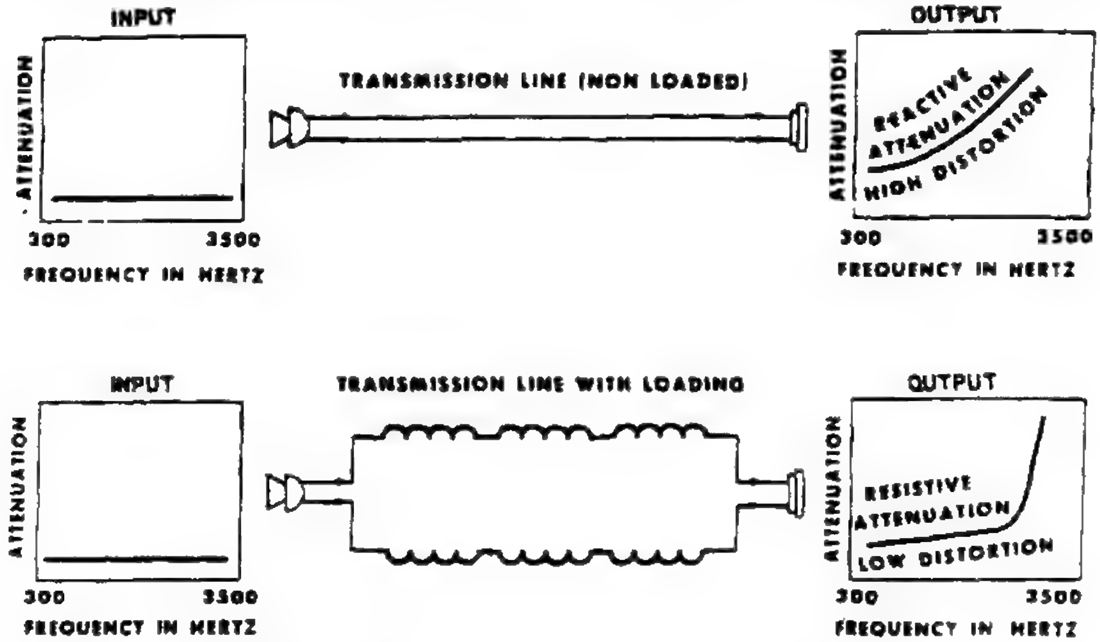


Figure 3. Loading reduces attenuation and distortion.

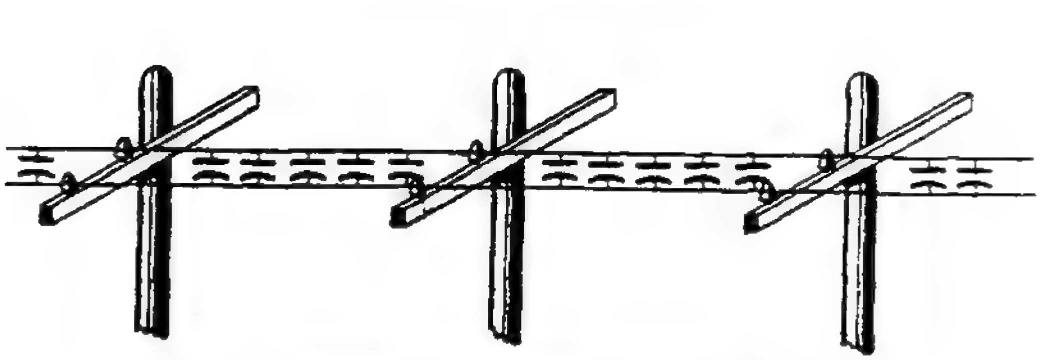


Figure 4. Capacitance is distributed along the entire line.

part of the line cannot counteract the capacitive effect distributed along the entire line. To prevent overloading and to counteract the distributed capacitance, two methods of loading are used.

a. Continuous Loading. Continuous loading consists of wrapping each wire of the pair to be loaded with a magnetic tape (fig 5). The tape, because it is magnetic, causes an inductive field to build up around each of the wires when current is flowing, counteracting the capacitive effect. Continuous loading is the best type of loading that can be used. But, because of its extremely high cost, it is impractical for most transmission lines and is used only on submarine (underwater) cables.

b. Loading Coils. Another method of loading is to insert small loading coils at frequent intervals along the length of the transmission line. The spacing must be such that there are several coils per wavelength of signal; otherwise the attenuation will increase rather than decrease. Loading coils are cheaper than continuous loading and this method is more frequently used.

6. LIMITATIONS OF THE LOADING METHOD

a. Although loading improves the transmission characteristics of a line, the method has limitations. It can only improve transmission to a certain degree; it only reduces line losses and distortion. And there is a limit to the number of loading coils that can be added in a transmission line circuit; too many coils cause loss of signal power and distortion.

b. The combination of inductance and capacitance causes the transmission line to act like a filter. Attenuation increases sharply when the frequency exceeds a certain value; this frequency is called the cutoff frequency. The cutoff frequency for a circuit depends upon the loading coils selected. Notice in figure 6 that the frequencies in the

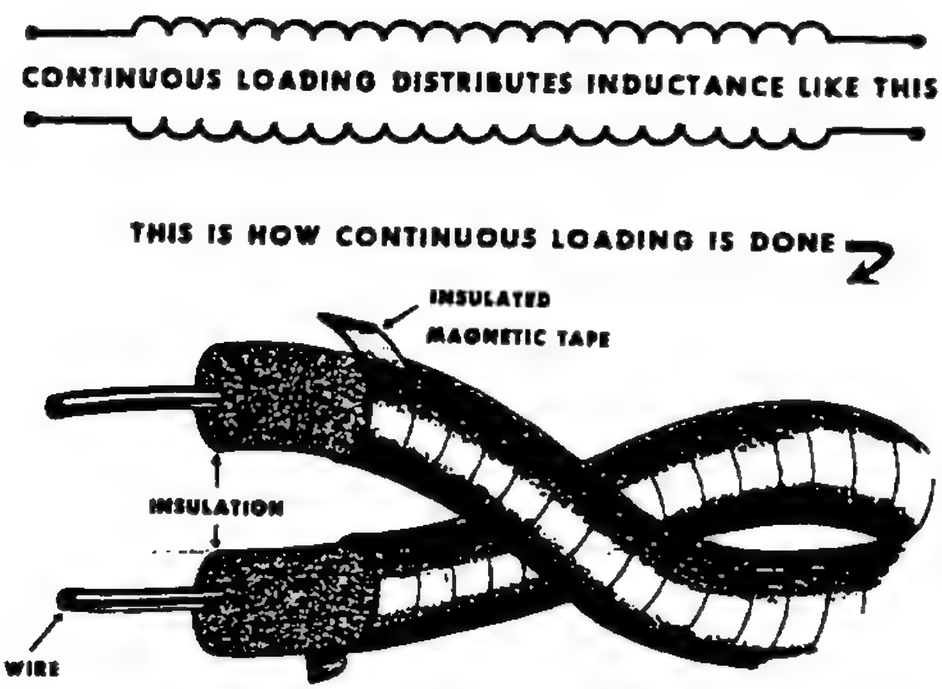


Figure 5. Continuous loading.

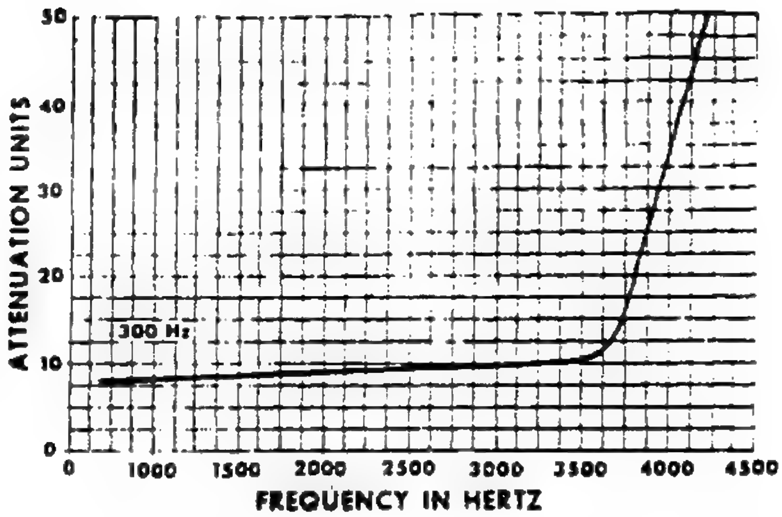


Figure 6. Loading affects frequencies transmitted.

300-3,500 Hertz range get about the same attenuation by the line. Above 3,500 Hertz however, the attenuation increases very sharply. The graph represents the response of a circuit designed to pass only voice frequencies.

7. SPIRAL-4 CABLE

a. Spiral-4 cable is used in many transmission line circuits. It has two pairs of wires (fig 7), is insulated with polyethylene, and is provided in standard 1/4-mile lengths. Spiral-4 cable may be used as a loaded or nonloaded line depending on the transmission line circuit requirements.

b. Each spiral-4 cable has a universal connecter at each end. To load a spiral-4 line, 6 millihenry coils are inserted at 1/4-mile intervals, between cable lengths. The cutoff frequency for spiral-4 cable is about 28,000 Hertz.

8. REPEATERS

a. The main job of a repeater is to amplify (increase) voice power. Repeaters are installed at various points along a line. When attenuated voice power passes through a repeater it is amplified; it gains extra power from the repeater.

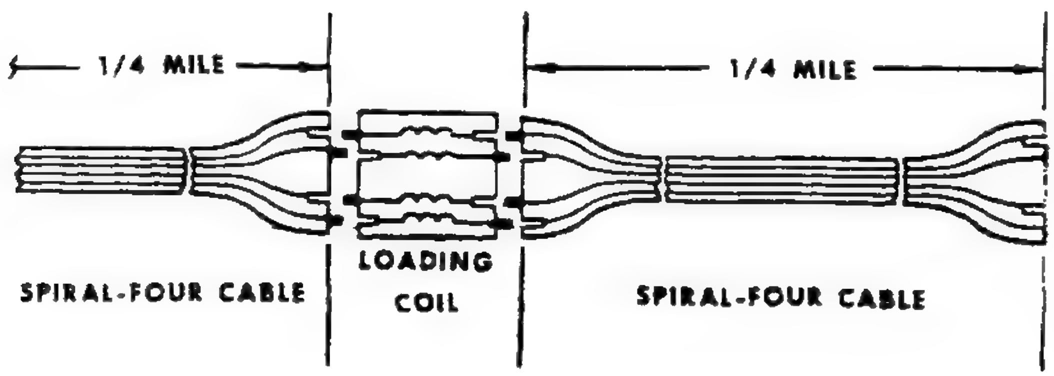


Figure 7. Loading in spiral-four cable.

b. You can see how this works in figure 8. Two transmission lines built from the same type and length of line are shown. A repeater has been installed in one of the lines. Notice that the input power to each line is the same (1 milliwatt). The output power, however, is different for each line. The power has been so greatly reduced in the line that has no repeater that no sound is produced in the receiver. In the other line, the power has been reduced the same amount. Yet at the receiver there is plenty of power left to produce audible sound. The repeater makes the difference. Although both lines give the same attenuation, the repeater has increased the power in one line and has overcome the attenuation.

c. Figure 8 also shows that the section of line extending from the transmitter to the repeater has reduced voice power from 1 milliwatt to 1/1,000 milliwatt by the time it enters the repeater. Inside, amplification takes place and the voice power comes out as 4 milliwatts. If you figured it out, you'd find that the repeater amplified the 1/1,000 milliwatt input 4,000 times. This is a tremendous gain of power and the part of a repeater that does this job is called an amplifier.

9. REPEATERS HAVE TWO AMPLIFIERS

Modern day repeater amplifiers may use electron tubes or transistors. The amplifiers are one-way devices which means that voice power can go through an amplifier in only one direction. Thus, each repeater requires two amplifiers: one for each direction of transmission. Figure 9 shows the symbol for a repeater. The amplifiers are represented by the arrow heads and the arrow heads point in the direction of transmission. The gain provided by the amplifier must overcome the loss caused by all of the other parts of the transmission system.

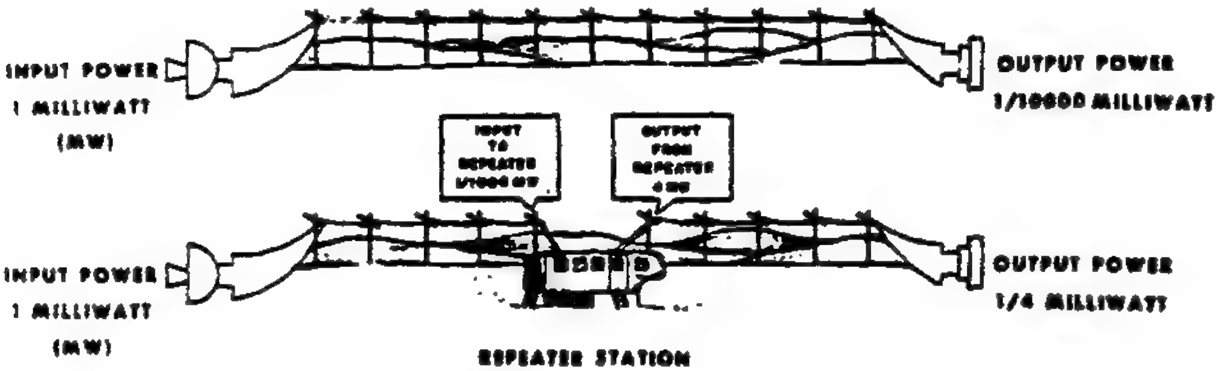


Figure 8. Repeater provides a gain.

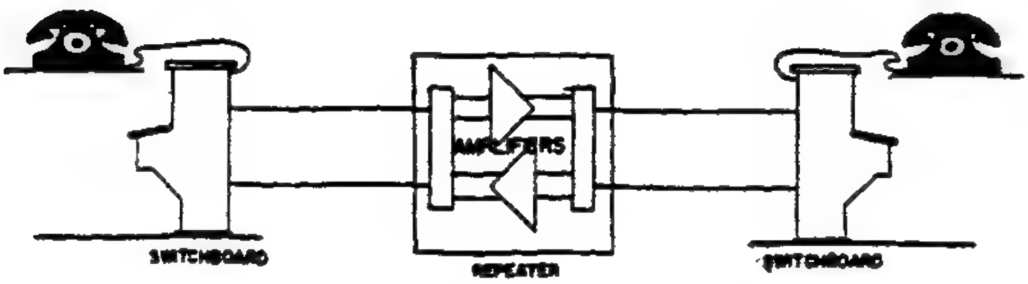


Figure 9. Repeaters have two amplifiers.

APPENDIX C

INTRODUCTION TO

COMPANDORS

SSTS 53105

OBJECTIVES:

1. To discuss principles of sound energy related to compandor operation.
2. To explain what a compandor is.
3. To show what a compandor does.
4. To explain how a typical volume compandor works.

INTRODUCTORY INFORMATION

In voice communications, interfering disturbances called noise and crosstalk constantly try to reduce intelligence to meaningless jumble. A special device which offers a practical method of dealing with these two main enemies of communication is the compandor.

In its important role, the compandor serves as a remedial device designed to transmit voice signals above the noise and crosstalk encountered in communication systems, thereby improving the quality of voice transmission. Refer to figure 1 on next page.

To understand how compandors offer relief from the disturbing effects of noise and crosstalk, you have to know something about how compandors work and about the basic principles of sound energy related to their operation. The purpose of this sheet, therefore, is to provide you with this information.

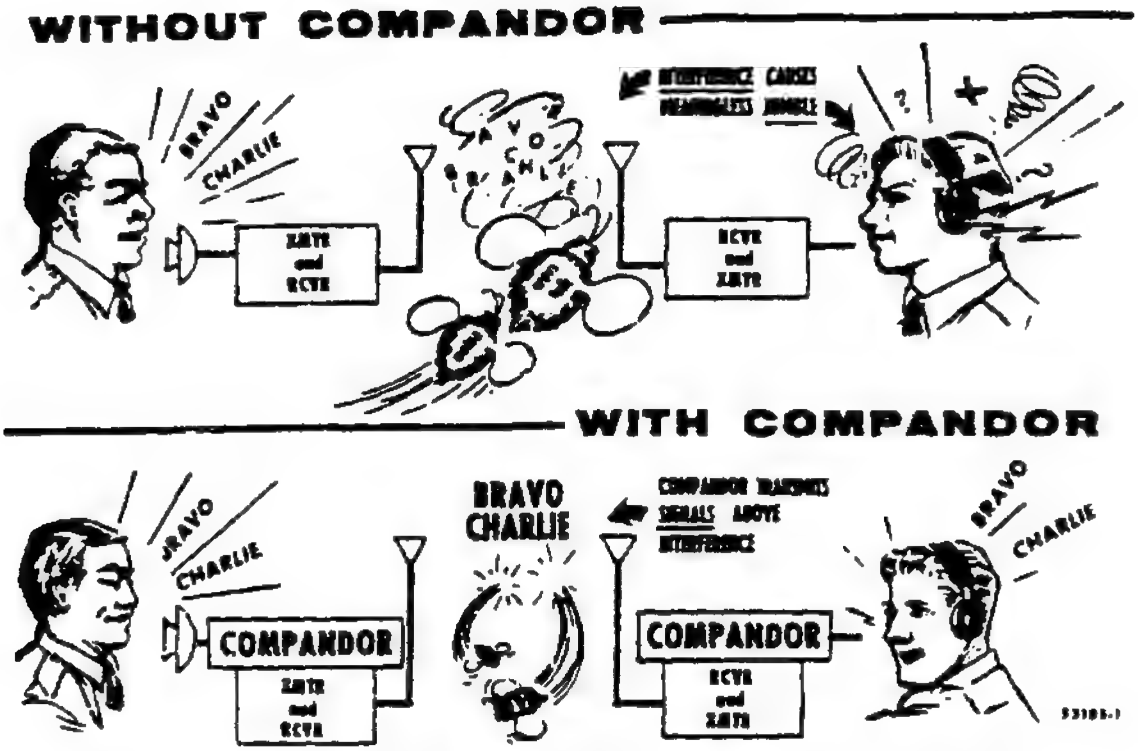


Figure 1. A compandor improves the quality of voice transmission.

Before we discuss compandors in this sheet, we'll first review some basic principles of sound (speech) energy related to compandor operation. Next we'll discuss the two main enemies of communication -- noise and crosstalk. Then we'll explain what a compandor is and show how it offers relief from noise and crosstalk. Next, we'll briefly discuss the different types of compandors and explain how a typical volume compandor works. Finally, we'll tell in what types of equipment compandors are used.

SOUND (SPEECH) ENERGY

In SSTS 53001, Basic Principles Of Sound-Review, you learned that your voice produces tones consisting of different frequencies and varying intensity. And you've learned that together these two signal characteristics, frequency and intensity, make up the speech energy that we convert into electrical energy by means of radio, telephone, or other communications systems. Because these characteristics of speech energy play an important part in the operation of compandors, let's review some other important facts about intensity and frequency.

FIRST, SPEECH FREQUENCY

Speech frequency ~~is the number of vibrations (Hertz) that make up~~ sound. The frequency range (Fig. 2) of human voice (male and female together) is from about 100 to 8,000 Hz. However, most of the energy of speech signals being transmitted is concentrated in a frequency range of 200 Hz to 3 200 Hz.

Remember, that in communication systems, we're not interested in transmitting all our vocal tones. We're interested in transmitting only enough frequencies so that a listener can understand what we say. (For additional information on frequency, review SSTS 53001.)

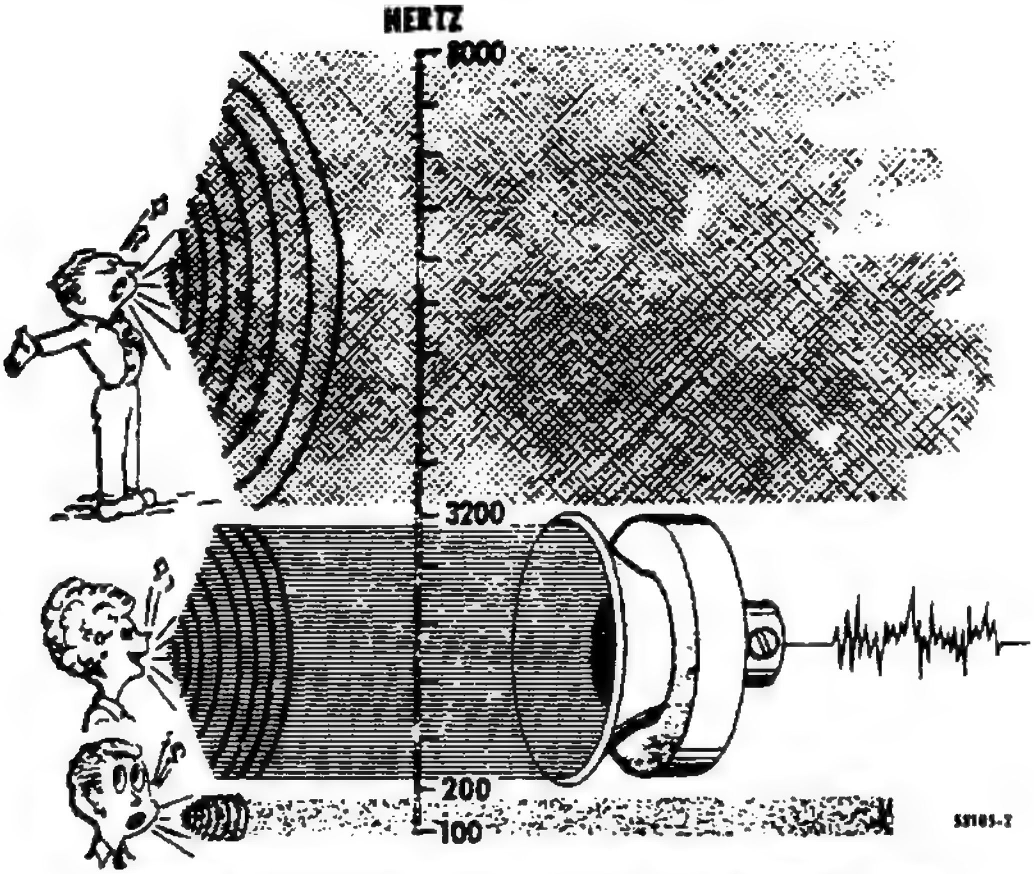


Figure 2. We only transmit part of the voice frequency range.

NEXT, SPEECH INTENSITY

Speech intensity refers to the volume, loudness, or power of speech. A common unit of measurement of intensity is the decibel (db).

The intensity or power level of our speech changes when we speak. Notice in A of figure 3 how this person's speech changes in power level. At point one, the power level is 10 db; at point two, it is 25 db; and so forth. At point four, the speaker reaches his highest power level of 50 db.

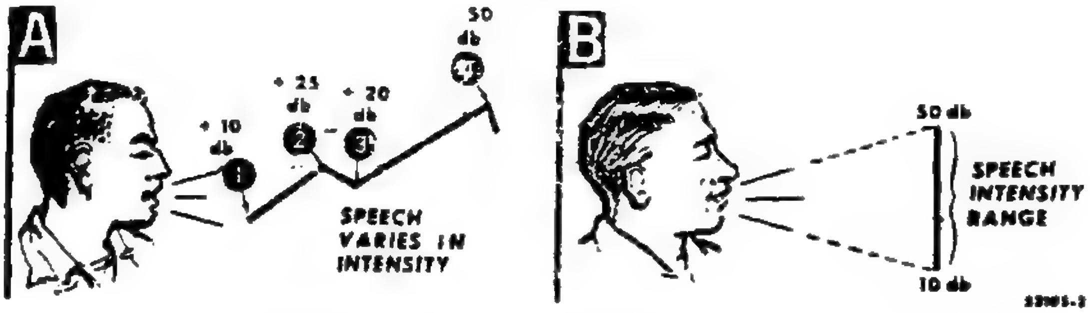


Figure 3. Our speech varies in intensity.

The difference between the speaker's lowest and highest power level shown in part B is 40 db (50 db - 10 db = 40 db). We call this difference the speech intensity range. (Refer to SSTS 53005, Power Losses and Gains -- Decibels, for complete information on db's.)

WHAT DETERMINES SPEECH INTENSITY RANGE?

Two main factors determine speech intensity range: the speaker and his words. Here's why.

A speaker can increase or decrease his speech intensity range by varying his tone of voice. For example, if he changes his voice from a whisper, when talking about "sweet nothings", to loud sounds when stressing a point, his intensity range will be greater than it will be if he speaks in the same tone.

Words also help to determine speech intensity range because our speech syllables produce different amounts of power. What is a syllable? Briefly, a syllable is one or more letters of a word taken together to form a sound. For example, the word America has four syllables (A-mer-i-ca). Each spoken syllable produces a different amount of power (see figure 4).

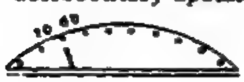
Figure 4. Words help determine intensity range.

Notice in figure 4 that the syllable "mer" produces the most power because the person stresses that syllable more than the others. Thus, "mer" helps to determine the power in the high end of the intensity range. Syllable "A" produces less power because it was stressed less. Therefore, "A" helps to determine the power in the lower end of the intensity range. Of course, the amount of power that syllables in this or any other word produce depends on the way the speaker stresses each syllable.

The normal intensity range for the average person is about 40 db; however, some people have an intensity range of 60 db or more.

TRANSMITTING A WIDE RANGE OF SPEECH POWERS

In communications, an operator's successive speech signals may differ in intensity by many decibels even though he may have an average intensity range. For example, when an operator speaks evenly at a uniform distance from the microphone, he establishes an average intensity of speech power  But if the operator deliberately shouts, stresses certain words or syllables, or moves closer to the microphone, the intensity increases . However, if the operator moves farther away from the microphone, or deliberately speaks in a quiet tone of voice, the intensity of speech power decreases.



Notice that the difference between the operator's loudest and weakest sounds recorded on the meters (page 4) is 60 db (80db-20db). This means that there is 1 million times as much power in the loudest sound as there is in the weakest sound because 60 db is equivalent to a power ratio of 1 million to 1. Transmitting such a wide range of speech powers presents some problems in communication systems.

PROBLEMS IN TRANSMITTING WIDE RANGE OF SPEECH POWERS

Two major problems encountered in transmitting wide ranges of speech power are as follows:

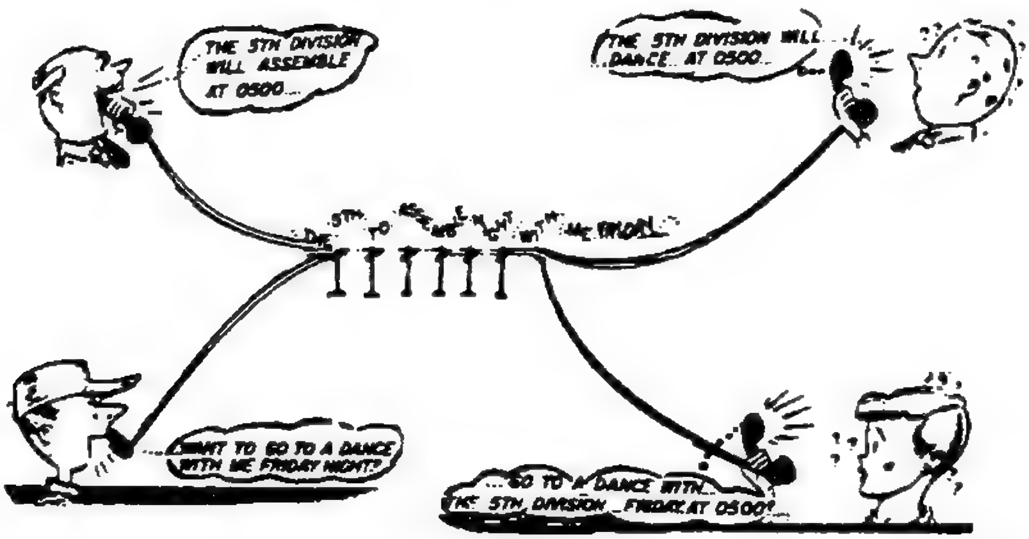
1. Low power (weak) signals become lost in electrical noise and crosstalk -- the two main enemies in communication systems.

2. High power (strong) signals constitute a major source for crosstalk.

Now what is crosstalk and noise? Let's find out by discussing each of these disorganizing forces separately.

FIRST, CROSSTALK

Crosstalk is interference (unwanted signals) that leaks from one channel to another in a communication system. When conductors that are close to each other carry messages as shown in figure 5, some signal power slips from one conductor to another, causing channel interference with the desired conversation. All of us have been exposed to crosstalk at one time or another while talking on the phone, listening to the radio, or watching TV.



53106-5

Figure 5. Crosstalk interferes with conversation.

Crosstalk may invade the privacy of a communication system at any time. But this type of interference is most noticeable and annoying during silent periods of a conversation, such as at the end of a sentence, between words, or during any other breaks in speech.

NOW, ELECTRICAL NOISE

Electrical noise is by far the principal enemy of communication and is the type of interference you'll be concerned with most in your work as a repairman. This type of interference is much more serious than crosstalk because it can come from any one of a number of sources. Besides, noise is generated afresh at almost every point in the transmission path. Now let's learn about electrical noise by answering these three questions:

1. What does electrical noise look like?
2. Where does electrical noise come from?
3. Why is electrical noise harmful to communications?

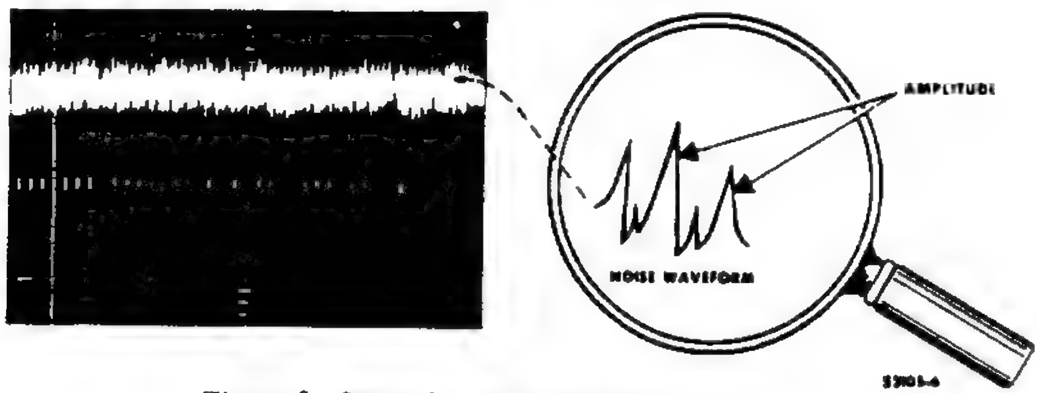


Figure 6. What electrical noise looks like.

FIRST, WHAT DOES ELECTRICAL NOISE LOOK LIKE?

Electrical noise takes on many shapes and forms. Figure 6 shows how electrical noise in communication channels may appear on an oscilloscope. Note that there is no particular pattern to the noise. The amplitude of the noise keeps changing; power level and frequency keep changing also.

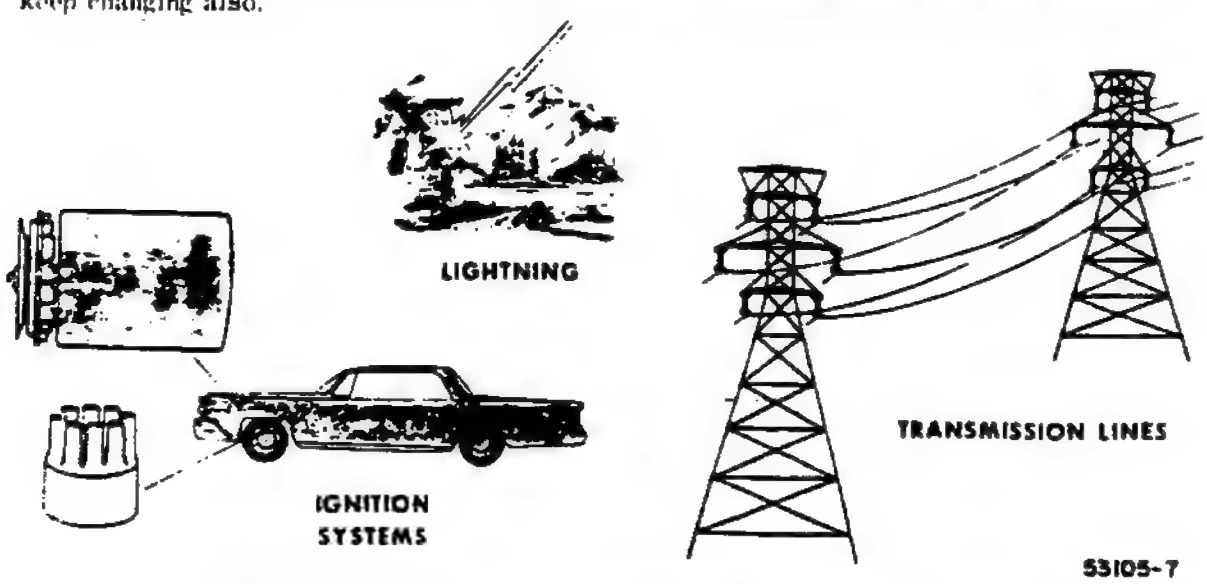


Figure 7. Sources of electrical noise interference from outside communication systems.

NEXT, WHERE DOES ELECTRICAL NOISE COME FROM?

Electrical noise interference comes from both outside and inside the communication system, however, most noise comes from outside the system. Outside noises originate from such sources as lightning, electrical-power transmission lines, and automobile ignition systems (see figure 7). These and similar sources radiate noise, in the form of electrical energy, in all directions; nearby communication systems pick up the radiated noise.

Noise within the communication system is generated by internal components of transmitting and receiving devices. You see, the flow of electricity through components causes electrons to bump into molecules within conducting material and, thus, set up noise. Notice that the movement of electrons in the resistor, shown in figure 8, generates a small electrical current (noise). You can also detect similar noise generated in tubes, transistors, and other components found within electronic equipment.

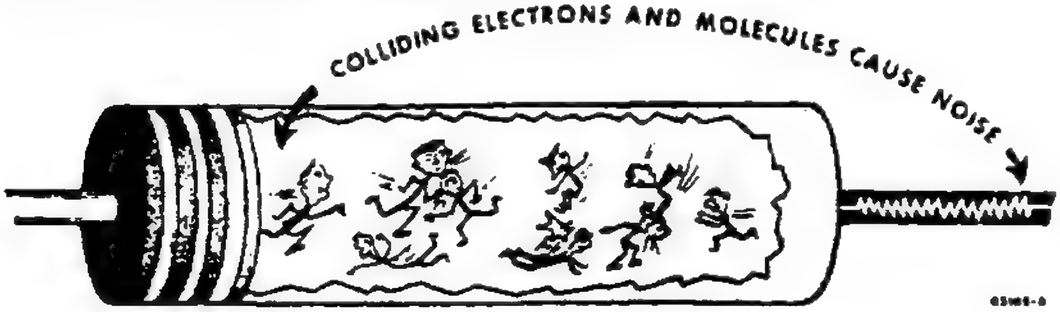


Figure 8. Where electrical noise interference from inside communications systems comes from.

FINALLY, WHY IS ELECTRICAL NOISE HARMFUL TO COMMUNICATIONS

The presence of noise in a communication system, regardless of where comes from, makes it difficult or impossible to hear the entire range of an operator's voice. You can hear the loudest parts of his voice because they are more powerful than the noise, but you cannot hear the quietest parts because of "masking." That is, the noise drowns out the weaker signals (see figure 9).

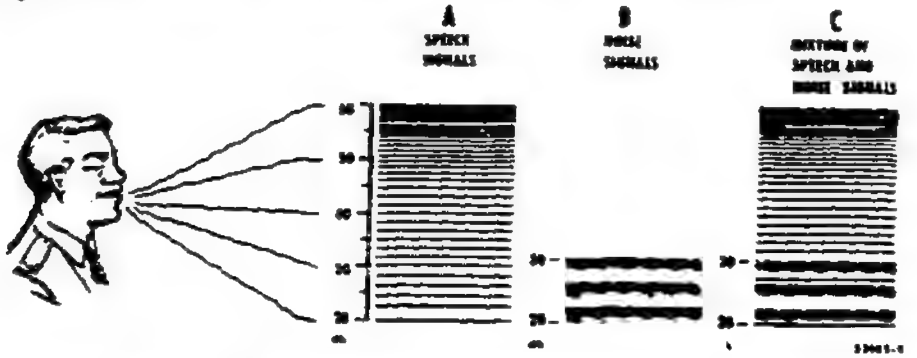


Figure 9. Noise drowns out weak speech signals.

Notice in A of figure 9 that the operator's speech signals extend over an intensity range of 40 db (60db-20db). Also notice in B of figure 9 that the assumed noise level in a communication system is 30 db. Now when the speech signals combine with the noise signals (C of figure 9), you'll notice that the noise level is sufficient to mask the operator's weak speech signals (signals below 30 db), thus drowning out parts of the operator's conversation.

Let's look at the "masking" effect that noise has on speech signals in another way.

ANOTHER LOOK AT THE WAY NOISE AFFECTS SPEECH SIGNALS

Another way to see how noise drowns out weak speech signals is to look at the shape of a speech signal before and after it has been exposed to the disorganizing force of noise (see figure 10).

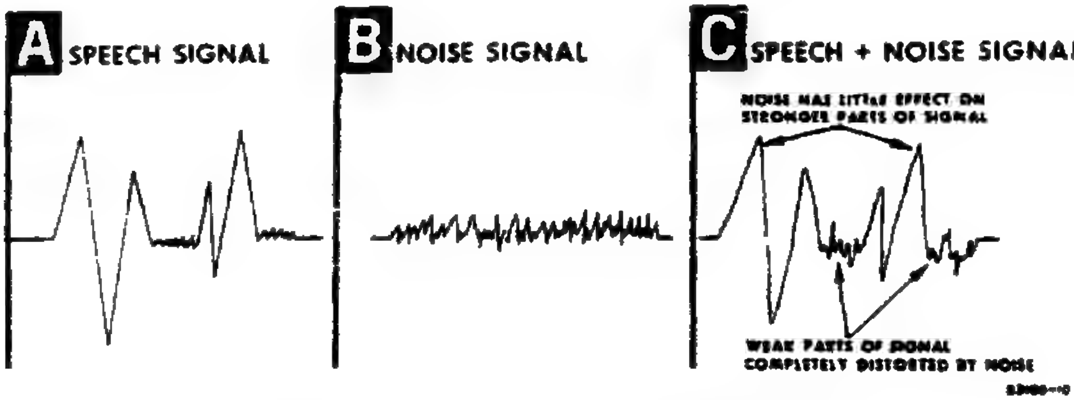


Figure 10. Noise distorts weak speech signals.

Refer to the shape of the speech and noise signals in parts A and B of figure 10. Now, look at part C and notice the shape of the speech signal after it mixes with the noise signal. Observe that the noise signal distorts the weaker parts of the speech signal. The reason -- the noise signal is more powerful than the weaker parts of the speech signal. In an actual communication system, the operator would not hear the weaker parts of this signal; instead he would hear noise plus the stronger parts of the signal.

You should also notice in part C that noise attacks the strong as well as weak parts of the speech signal. However, because the strong parts of the signal are so much more powerful than noise, noise has little or no effect on strong signals. Because noise has little effect on strong signals, the remainder of this text will deal with the effect of noise and crosstalk on weak signals and not on strong signals.

WHAT WE CAN DO ABOUT INTERFERENCE FROM NOISE AND CROSSTALK

Now that you've learned what noise and crosstalk are and how they affect communications, what can we do about them? It's impossible to get rid of these interferences entirely because, as you've already learned, they can creep into communication systems at any time. It's possible, however, to reduce their effects by improving the signal-to-noise ratio. Let's see what this means.

IMPROVING THE SIGNAL-TO-NOISE RATIO

The signal-to-noise ratio is a comparison of signal strength to noise strength. This ratio, usually stated in db, tells how much greater the signal power is above the noise power. For example, if there is 10,000 times as much power in a certain speech signal compared to the noise produced ~~in a communication system~~, the signal-to-noise ratio is 10,000 to 1 or 40 db (see A of figure 11). But if there is only 10 times as much power in the signal compared to noise, the signal-to-noise ratio is 10 db (see B of figure 11).

If we assume that the noise level is constant, and the signal-to-noise ratio becomes smaller, then obviously, the power in the speech signal also becomes smaller. If this is allowed to go too far, the weak signals will become lost in the noise (and crosstalk), thus destroying communications.

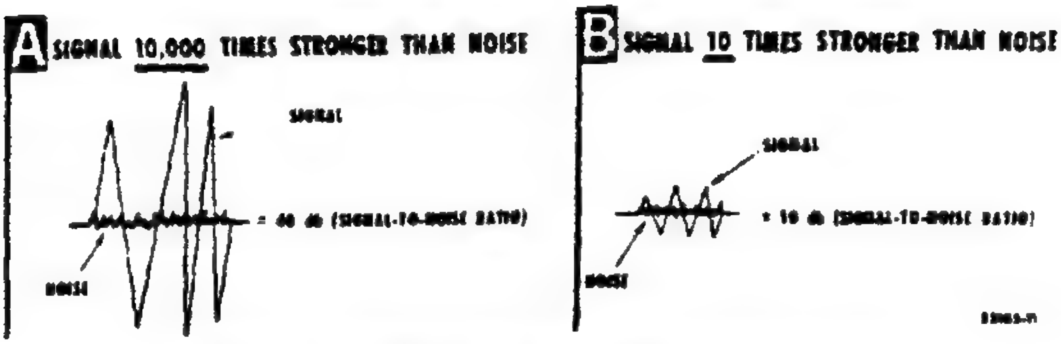


Figure 11. Comparing signal strength to noise strength.

To prevent noise and crosstalk from masking weak signals in a communication system, we must increase the signal-to-noise ratio. That is, we must increase the power of the weak signals above that of the noise and crosstalk. How can we do this? By using a special device called a COMPANDOR.

What is a COMPANDOR? We'll see right after we briefly review the information that we've discussed so far.

REVIEW EXERCISE

Answers to this review exercise are on page 20.

The test items below will help you review the information you have learned about sound (speech) energy, noise, crosstalk, and signal-to-noise ratio. Read each item carefully and place your answer in the space(s) provided.

1. The two signal characteristics that make up speech energy are ____________________ and ____________________.

2. What is the frequency range of most speech signals that are transmitted?

________ ________________.

3. Some other terms for speech intensity are ____________________, ____________________ and ____________________

4. The difference between a speaker's lowest and highest speech power level is called

____.

5. The amount of power that a syllable produces when spoken depends upon

____.

6. List two major problems encountered in transmitting wide ranges of speech power.

____

____

7. The principal interfering disturbance in voice communication is called ____________

8. Briefly -- what is "masking"? ____

____

9. One way to reduce the ill effects from noise and crosstalk is to improve the __________

____

IN GENERAL, WHAT IS A COMPANDOR?

A compandor is an electronic device consisting of two separate voice-operated units. One unit is an intensity range <u>COMpressor</u>; the other is an intensity range <u>exPANDOR</u>. By combining parts of each name we arrive at the name, COM-PANDOR. See figure 12.

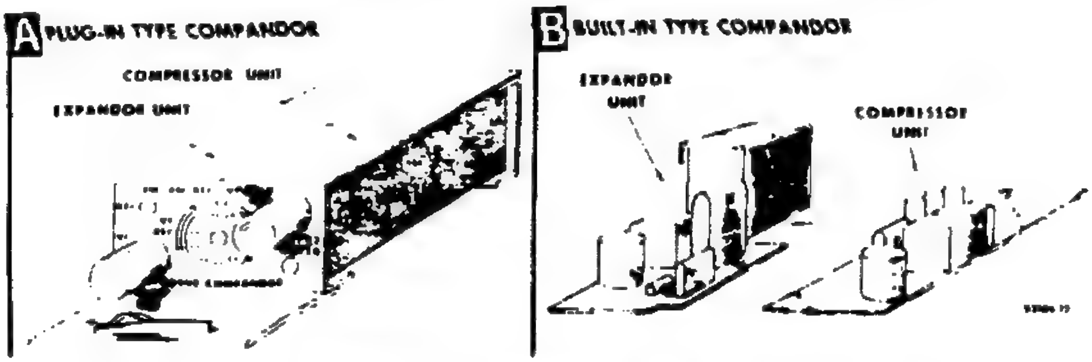


Figure 12. A compandor is a combination intensity range COMpressor and exPANDOR.

In actual equipment, you'll find that the compressor and expandor units may be combined as a single plug-in type unit as shown in A of figure 12, or they can be separate units built permanently in the equipment as shown in B of figure 12. However, regardless of whether the units are together or separate, you should remember that a compandor consists of both a compressor and an expandor.

IN GENERAL, WHAT DOES A COMPANDOR DO?

A compandor improves the quality of voice transmission by decreasing the effects of interference from noise and crosstalk. See figure 13.

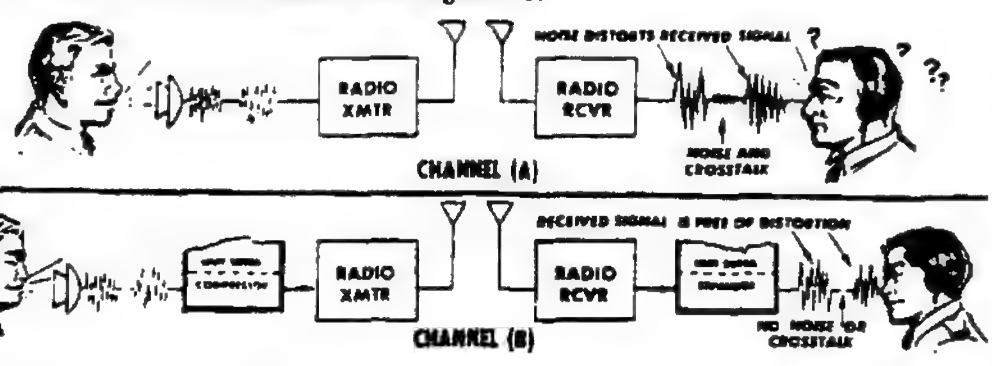


Figure 13. Comparison of a communication channel with and without a compandor.

Notice in channel A of figure 13 that the listener at the receiver end of the channel is confused. The reason -- he isn't getting the complete message because of interference from noise and crosstalk. Notice that not only has noise distorted the received signal, but noise and crosstalk have crept in between words, confusing the listener further.

Now look at channel B of figure 13. Notice that the listener in this channel is not confused: he hears everything the other fellow is saying. Why? Because the compandor (compressor and expandor) decreases the interference resulting from noise and crosstalk. Now the received signal is free from distortion, and just as important there is no noise or crosstalk between the words.

Before we discuss the details of how compandors work, first notice the physical location of the compandor units as shown in channel B of figure 13. Observe that the compressor is located at the transmitting end of the communication channel, while the expandor is located at the receiving end of the same channel. In the equipment that you'll repair, you'll find this the normal arrangement of compandor units. Now, let's discuss how compandors work.

HOW COMPANDORS WORK (GENERAL)

You've learned that compandors are remedial devices offering relief from the effects of noise and crosstalk. But how? To find out, you'll have to study the two units that make up the compandor: the compressor unit and the expandor unit.

FIRST, THE COMPRESSOR UNIT

A compressor unit performs the following two important functions:

1. It raises the power level of weak signals (increases signal-to-noise ratio) so weak signals can be transmitted above the noise and crosstalk encountered in the system between the compressor and expandor.

2. It attenuates (cuts down) very strong signals to help prevent crosstalk from spreading to other communication channels.

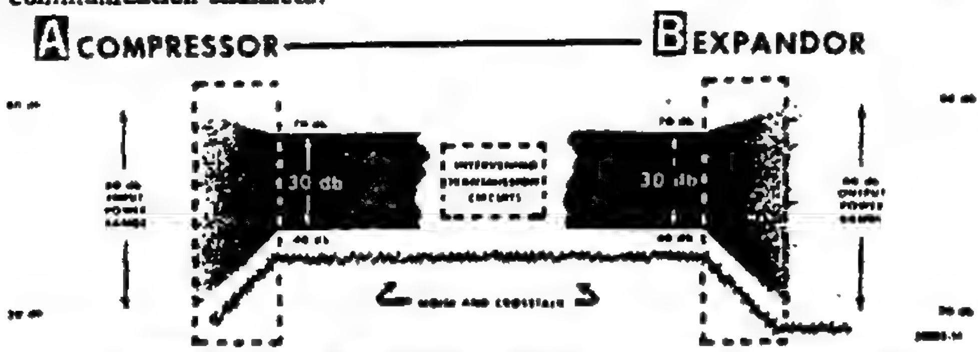


Figure 14. The overall action of a compressor and expandor.

A compressor accomplishes these functions by automatically compressing (reducing) the intensity range of speech signals before transmitting the signals. Notice in A of figure 14 that the input power range to the compressor is 60 db, but the output range is 30 db. This change in range is the result of a shift in the amount of power imparted to the speaker's sounds. Look closely and you will see that the weakest sounds increased in power (from 20 db up to 40 db), while the very strongest sounds decreased in power (from 80 db down to 70 db). In other words, the input power range was compressed by 30 db. Now remember we are transmitting the same sounds but the sounds transmitted are at different power levels from the original power levels. The power that weak signals gain through compression helps them to overcome the noise and crosstalk encountered between the compressor and expandor. The power that the strong signals lose serves to cut down on crosstalk to other channels.

NOW, EXPANDOR UNITS

An expandor unit works opposite to the compressor unit. That is, the expandor restores the reduced power range back to its original range once the speech signals reach their destination. How does the expandor do this? Let's see.

Notice that the power range at the input to the expandor (shown again in figure 15) is 30 db, but the output power range is 60 db. This change in power range is the result of another shift in the amount of power imparted to the speech sounds. However, note that the shift in power is opposite to that shown in the compressor. That is, the weakest sounds decreased in power (from 40 db down to 20 db), while the strongest sounds increased in power (from 70 db up to 80 db). In other words, the expandor restored the 30 db power range, produced by the compressor, back to its original range of 60 db.

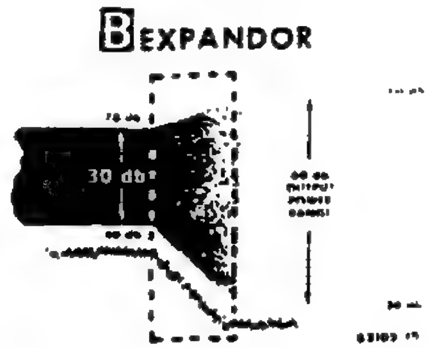


Figure 15. Action of an expandor.

Now how do compandors decrease the effects of noise and crosstalk in communication systems? By simply raising the power level of weak signals (increasing the signal-to-noise ratio) before the signals are exposed to channel interference.

It is important for you to know that a compandor does not eliminate noise and crosstalk, but offers relief from their effects. This is why we call the compandor a remedial device.

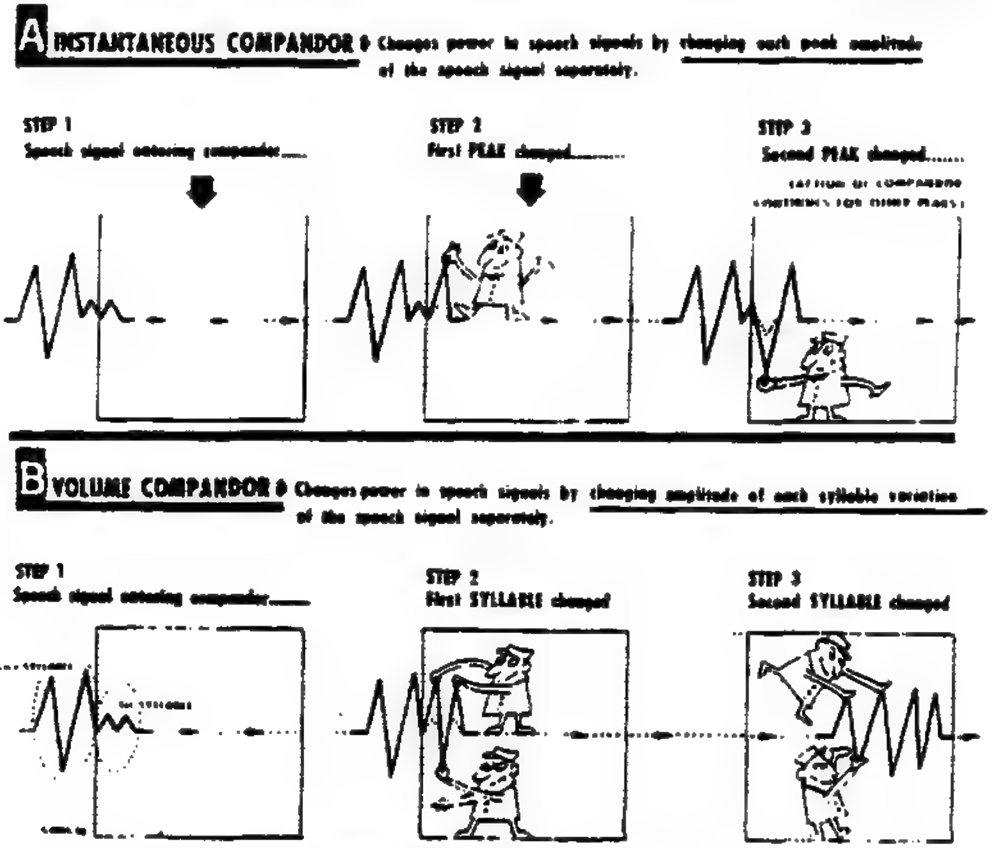


Figure 16. Compandors differ in the way they change power in speech signals.

BASIC TYPES OF COMPANDORS

Basically, there are two main types of compandors: a volume (or syllabic) compandor and an instantaneous compandor. The main difference between the two compandors is the way they change the power of speech signals. To get an idea how the two compandors vary the power of speech signals, study the steps outlined in A and B of figure 16.

In figure 16, notice that the input speech-signal to both compandors is the same; the only difference is the way each compandor changes the power level of the speech signal. Be sure you understand that the instantaneous compandor changes the power by changing each instantaneous peak amplitude of the signal. The volume compandor, however, changes the power in the signal by changing the amplitude of each syllabic variation (rather than each individual peak). Remember that both types of compandors do the same job, but in a different way.

In the remainder of this text, we will discuss only the volume compandor because this is the type used in the transmission system you'll be repairing. The instantaneous compandor is used in pulse-code modulation (PCM) transmission systems.

FIRST, A BLOCK DIAGRAM OF A TYPICAL VOLUME COMPANDOR

The volume compandor, like other compandors, combines both a compressor and an expandor. Each of these units (often called companding units) contains a VARIABLE LOSS DEVICE, AMPLIFIER, and RECTIFIER CONTROL CIRCUIT. Locate these component parts in the compressor and expandor units shown in figure 17.

Figure 17. Block diagram of a typical volume compandor.

The three component parts you located in each of the companding units are responsible for changing the intensity range of speech signals. The parts in the compressor work together to decrease the intensity range, while similar parts in the expandor work together to increase the intensity range.

Now that you know the names of the component parts in the companding units, let's briefly discuss the purpose of each part.

1. Variable Loss Device -- determines the overall gain of each syllabic variation of the input signal. In the compressor unit, the variable loss device attenuates strong signals (syllables) more than it does weak signals (syllables); but in the expandor unit, it attenuates weak signals more than it does strong signals.

2. Amplifier -- increases the power of all signals that leave the unit.

3. Rectifier Control Circuit -- controls the amount of signal (syllable) attenuation that takes place in the variable loss device.

You should notice that the component parts in the expandor are connected in reverse to the way the parts are connected in the compressor. The reason -- it is necessary for the expandor to reverse the action of the compressor.

NEXT, HOW A TYPICAL VOLUME COMPANDOR WORKS

To understand specifically how a volume compandor works, you need to know how the component parts you just learned about work together in each companding unit. A good way to learn how the parts work together is to trace the signal path through the compressor unit and then trace the signal path through the expandor unit.

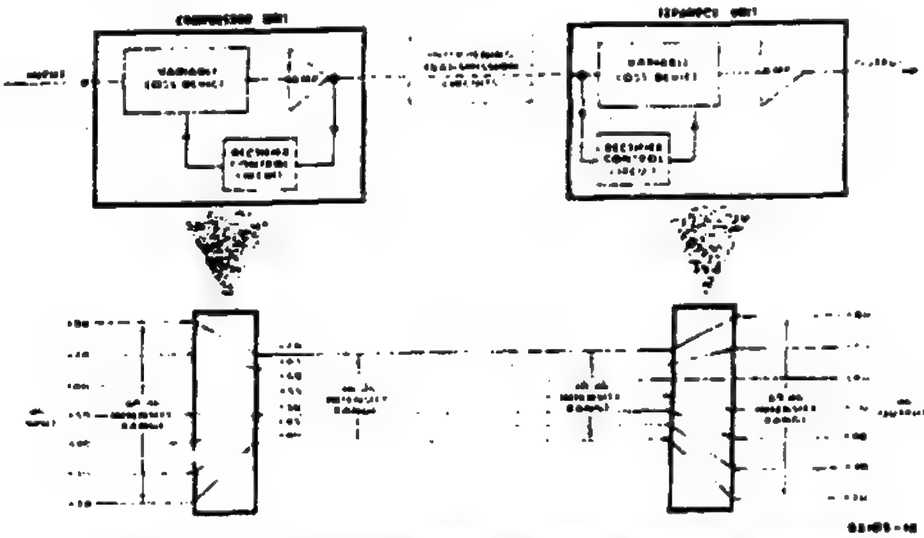


Figure 18. The overall action of a typical volume compandor.

FIRST, THE COMPRESSOR UNIT

As you read, follow the arrows in the compressor unit of figure 18. A speech signal enters the compressor and passes through the variable loss device and then through the amplifier. A part of the output signal from the amplifier goes to the rectifier control circuit where it changes to a control current. In turn, this control current goes to the variable loss device that controls the amount of input signal attenuation. Although all this may sound complicated, you'll see as you read further, that it isn't so complicated after all.

It is important for you to understand that the level of the control current varies directly with the strength of the incoming speech signals. That is, if a weak speech signal is present at the variable-loss device, the control current is small and the attenuation of the weak signal is very low (your eye cannot detect the small amount of attenuation). However, if a strong speech signal is present, the control current is high and the attenuation of the signal is high.

In other words, if a signal from a two-syllable word coming into the variable loss device looks like this  , when it leaves the device and goes to the amplifier, it will look like this  . Notice that the variable loss device attenuates the strong part of the signal more than the weak part. This attenuation of the strong signals results in the compression of the high end of the intensity range.

Now, how does the low end of the intensity range become compressed? This is accomplished by not attenuating the weak signals too much before sending them to the amplifier where they increase in power.

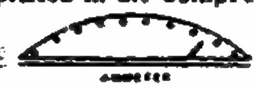
Another name for the compression action, performed by the component parts in the compressor, is companding action.

Refer to figure 18 and note the results of the companding action of the compressor on the different power levels of the 60 db input intensity range. Notice that each input power level below 60 db increases in power while each power level above 60 db decreases in power. Through this companding action, the intensity range is compressed from 60 db to 30 db. This amount of compression is sufficient to overcome the effects of noise and crosstalk that would be encountered between the companding units.

NOW, THE EXPANDOR UNIT

Follow the arrows in the expandor unit of figure 18, as you did in the compressor unit, and remember that the action of the expandor is just opposite to that of the compressor.

A portion of the input signal from the compressor enters the expandor and goes to the rectifier control circuit where it is converted to a control current. In turn, this control current goes to the variable loss device that controls the amount of signal attenuation.

Now, the level of the control current in this unit varies inversely with the strength of the incoming speech signal. That is, if a weak signal  (amplified in the compressor) is present at the variable loss device, the control current is large  and the attenuation of the signal in the variable loss device is high  . But, if a strong signal  is present at the variable loss device, the control current is small  and the attenuation of the signal is low  . In other words, if a signal from a two-syllable word coming into the variable loss device looks like this  when it leaves the device and goes to the amplifier, it will look like this  . Notice that the variable loss device attenuates the weak part of the signal more than the strong part. Attenuating weak signals more than strong signals expands the compressed intensity range back to its original range.

In figure 18 notice the results of the companding action of the expandor on the different power levels of the 30 db intensity range. Each power level above 60 db has increased in power while each power level below 60 db has decreased in power. The net result is that the intensity range has expanded from 30 db back to the natural power range of 60 db.

NOW, CHARACTERISTICS THAT CONTROL THE PERFORMANCE OF VOLUME COMPANDORS

The characteristics that control the performance of a compandor are compression-expansion ratio, companding range, and attack and recovery times. We'll discuss each of these characteristics separately starting with compression-expansion ratio.

FIRST, COMPRESSION-EXPANSION RATIO

A compression and an expansion ratio represents the degree to which speech signals are compressed and expanded. Both ratios are expressed (in db) with figures based on the relationship of input to output power ranges. See figure 19.

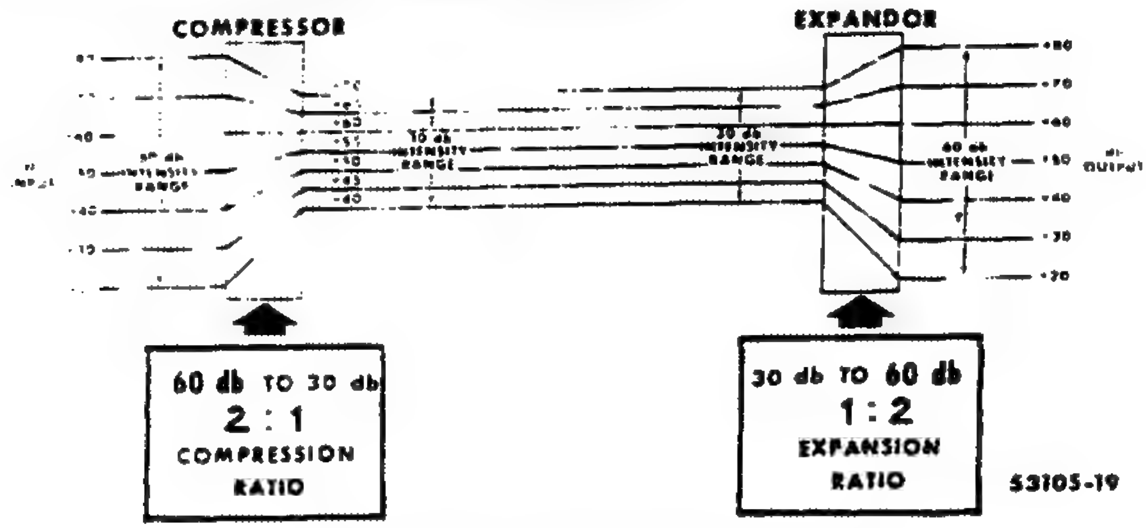


Figure 19. Compression and expansion ratios.

Notice in figure 19 that the input intensity range to the compressor is 60 db, but the output power range is 30 db. The compression ratio, therefore, is 60 db to 30 db or 2:1. A compression ratio of 2:1 means that the speech energy, traveling in the system between the compressor and expandor, has an intensity range of one-half its original value.

Now, look at the expandor unit of figure 19. Notice that the input intensity range to the expandor is 30 db, but the output power range is 60 db. Therefore, the expansion ratio is 30 db to 60 db or 1:2. An expansion ratio of 1:2 means that the intensity range leaving the expandor is twice as great as the intensity range entering the expandor. Thus, the expansion ratio is just the reverse from that of the compression ratio.

You'll find that in most volume compandors, the compression ratio is 2:1 and the expansion ratio is 1:2. The reason -- a compression ratio greater than 2:1 can excessively distort speech signals, while a smaller compression ratio will not sufficiently improve the signal-to-noise ratio for weak signals.

NEXT, THE COMPANDING RANGE

The companding range is the intensity range over which companding action occurs. The companding range of most compandors is usually between 50 db and 60 db. A companding range between these two db limits is usually large enough to provide proper signal-to-noise improvement over the wide intensity range of speech signals. Any high or low speech signal that may appear outside the 50 db to 60 db range is limited without affecting communication messages too much. Notice in figure 19 that companding action takes place over a companding range of 60 db.

HOW, ATTACK AND RECOVERY TIMES

You've learned that the gain or loss imparted to speech signals in a volume compandor is controlled by syllabic variations of the input signal and not by individual speech peaks. Now to make sure that syllabic variations control the gain, time constants are designed into the control circuits of both the compressor and expandor. These time constants are called attack and recovery times.

The attack time is the time it takes to change the power of a speech signal from a low power value to a higher power value. The recovery time is the time it takes to change the power of a speech signal from a high value to a lower value. Normal values for these times are about 3 milliseconds for the attack time and about 13.5 milliseconds for the recovery time.

The attack and recovery times must be properly selected to avoid distorting the speech signals. For example, if the attack time is too slow (less than 3 milliseconds), parts of the speech syllables will be distorted. If the recovery time is too slow, the expandor will not give the proper expansion between syllables. In addition to selecting the proper time constants, it is necessary to synchronize the attack and recovery times of the compressor and expandor. That is, the action of the expandor must follow the reverse action of the compressor, or the speech signals will also be distorted.

FINALLY, WHERE ARE COMPANDORS USED

You'll find compandors used in many types of equipment to reduce the interfering effects of noise and crosstalk. Some specific types of equipments that utilize compandors are telephone carrier equipment, radio equipment, and pulse-code modulation (PCM) multiplexing equipment.

FINAL SUMMARY

In this sheet, you first studied the principles of sound energy related to compandor operation, then you learned what a compandor is, what it does, and how a typical volume compandor works. Now test yourself on how much you have learned by answering the following review items.

Correct answers to these items are given on pages 20 and 21.

REVIEW ITEMS

1. What is the intensity range for a speaker whose lowest power level of speech is 20 db and highest power level is 70 db?

ANSWER: ____________________

2. What other factor, besides words, determines intensity range?

ANSWER: ____________________

3. Briefly explain how words help to determine intensity range?

ANSWER: ____________________

____

4. Briefly explain what is meant by crosstalk.

ANSWER: ____________________

____

5. List the chief sources of electrical noise.

ANSWER: ______________________________

____

6. Briefly explain how noise affects communications.

ANSWER: ______________________________

7. Briefly explain what effect noise has on strong speech signals.

ANSWER: ______________________________

8. Explain the meaning of a 40 db signal-to-noise ratio.

ANSWER: ______________________________

9. To decrease the ill effects of noise and crosstalk, the signal-to-noise ratio for weak signals must be (increased) (decreased). (Select one.)

10. What two units make up a compandor?

ANSWER: ______________________________

11. What is the purpose of a compandor?

ANSWER: ______________________________

____

12. Physically, where is the compressor unit usually located in a communication system?

ANSWER: ______________________________

13. List the two functions of a compressor unit.

ANSWER: ______________________________

____

14. How does a compressor unit reduce the intensity range?

ANSWER: ______________________________

15. What is the main function of an expandor unit?

ANSWER: ______________________________

____

16. Briefly explain the difference between instantaneous and syllabic compandors.

ANSWER: ______________________________

____

17. List the component parts of a typical volume compandor and give the purpose of each part.

ANSWER:____

18. What is meant by a compression-expansion ratio?

ANSWER:____

19. Explain the meaning of a compression ratio of 2 to 1.

ANSWER:____

20. Briefly explain the meaning of a 50 db companding range.

ANSWER:____

21. Briefly explain what is meant by attack time?

ANSWER:____

22. List the types of equipment in which compandors are used.

ANSWER:____

ANSWERS TO REVIEW EXERCISE ON PAGE 9

1. Frequency and Intensity.

2. 200 Hz to 3,200 Hz.

3. Volume, loudness, and power.

4. Speech intensity range.

5. The stress placed upon the syllable by the speaker.

6. Low power signals become lost in electrical noise. High power signals are a source of crosstalk.

7. Noise.

8. The drowning out of weak signals by noise.

9. Signal-to-noise ratio.

ANSWERS TO FINAL REVIEW ITEMS STARTING ON PAGE 17

1. 50 db.

2. The speaker.

3. Different amount of power is produced by syllables of words when spoken. In turn this difference in power determines the intensity range.

4. Crosstalk is the result of interfering signals "leaking" from one communication channel to another.

5. Lightning, power transmission lines, automotive ignition systems, resistors, transistors, and tubes.

6. Noise drowns out weak speech signals.

7. Noise has little or no effect on strong signals.

8. The signal is 10,000 times as strong as the noise.

9. Increased.

10. Compressor and expandor.

11. A compandor increases the quality of voice transmission by decreasing the effects of interference from noise and crosstalk.

12. In the transmitter section.

13. A compressor raises the power (increases signal-to-noise ratio) of weak signals, and attenuates strong signals.

14. By automatically compressing all speech signals before transmitting the signals.

15. The main function of an expandor unit is to restore the compressed intensity range back to its original range.

16. Instantaneous compandors change the power in speech signals by changing each instantaneous peak value of a speech signal. A syllabic compandor changes the power in speech signals by changing syllabic variations.

17. Variable Loss Device -- determines the overall gain of each syllabic variation of the input signal.

Amplifier -- increases the power of all signals that leave the unit.

Rectifier Control Circuit -- controls the amount of signal attenuation that takes place in the variable loss device.

18. A compression-expansion ratio is a figure which represents the degree to which speech signals are compressed and expanded.

19. A 2 to 1 compression ratio indicates that the speech energy traveling between the compressor and expandor has an intensity range equal to one-half its original value.

20. A 50 db companding range is the intensity range over which companding, or compressing action, takes place.

21. Attack time is the interval of time it takes to change the power of a speech signal from a low power value to a higher power value.

22. Carrier, radio, and pulse-code multiplexing equipment.